Panasonic

Hybrid IP-PBX

Feature Guide

Model KX-TDA30/KX-TDA100/KX-TDA200

Thank you for purchasing the Panasonic Hybrid IP-PBX, KX-TDA30/KX-TDA100/KX-TDA200.
Please read this manual carefully before using this product and save this manual for future use.

KX-TDA30/KX-TDA100/KX-TDA200: Version 1.0

Introduction

About this Feature Guide

This Feature Guide is designed to serve as an overall feature reference for the Panasonic Hybrid IP-PBX.
It explains what this PBX can do, and how to obtain the most out of its many features and facilities.

This manual contains the following sections:

Section 1, Call Handling Features

Provides details about the call handling features.

Section 2, System Configuration and Administration Features

Provides details about the system configuration and administration features.

Section 3, Programming

Provides system programming instructions.

Section 4, Appendix

Provides tables listing capacity of system resources, and tones and ring tones. It also provides the list of abbreviations.

Index

Provide feature titles, important words to help you access the required information easily.

Terms used in this Feature Guide

Installation Manual References

The required installation instruction titles described in the *Installation Manual* are noted for your reference.

Feature Guide References

The related feature titles described in this *Feature Guide* are noted for your reference.

User Manual References

The operation required to implement the feature described in the *User Manual* is noted for your reference.

Abbreviations

There are many abbreviations used in this manual (e.g., "PT" which stands for proprietary telephone). Please refer to the list in this manual (→ 4.3 List of Abbreviations) for the meaning of each abbreviation.

About the other manuals

Along with this Feature Guide, the following manuals are available to help you install, and use this PBX:

Installation Manual

Provides instructions for installing the hardware and maintenance of the PBX.

User Manual

Provides operating instructions for end users using PTs, SLTs, PSs, or DSS Consoles.

CE

The KX-TDA30E, the KX-TDA30NE, the KX-TDA30GR, and the KX-TDA30CE are designed to interwork with the:
- Analogue Public Switched Telephone Network (PSTN) of a European country
- Pan-European Integrated Services Digital Network (ISDN) using ISDN basic rate access

The KX-TDA100E/KX-TDA200E, the KX-TDA100NE/KX-TDA200NE, the KX-TDA100GR/KX-TDA200GR, and the KX-TDA100CE/KX-TDA200CE are designed to interwork with the:
- Analogue Public Switched Telephone Network (PSTN) of a European country
- Pan-European Integrated Services Digital Network (ISDN) using ISDN basic rate access
- Pan-European Integrated Services Digital Network (ISDN) using ISDN primary rate access
- ONP 2048 kbit/s digital structured leased lines (D2048S)

We, Panasonic Communications Co., Ltd./Panasonic Communications Company (U.K.) Ltd., declare that this equipment is in compliance with the essential requirements and other relevant provisions of Directive 1999/5/EC.
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Notes

- There are some optional service cards, PTs, and features which are not available in some areas. Additionally, there are some optional service cards and features for the KX-TDA100/KX-TDA200 which are not available for the KX-TDA30, or vice versa. Please consult your certified Panasonic dealer for more information.
- Displays are described in English as samples.
- While all system programming can be performed through PC programming (→ 2.3.1 PC Programming), PT programming can only cover a subset (→ 2.3.2 PT Programming). In Section 1 Call Handling Features and Section 2 System Configuration and Administration Features, programming references such as "→ Date & Time [000]" indicate that system programming can be done by PT programming.

For further details, please refer to the on-line help of the Maintenance Console (→ 3.2.1 Installing and Starting the Maintenance Console).

Feature Highlights

Networking Features

This PBX supports the following networking features:

TIE Line Service

A TIE line is a privately leased communication line between two or more PBXs, which provides cost effective communications between company members at different locations. (→ 1.27.1 TIE Line Service)

Virtual Private Network (VPN)

VPN is a service provided by the telephone company. It uses an existing line as if it were a private line. (→ 1.27.2 Virtual Private Network (VPN))

QSIG Network

QSIG is a protocol which is based on ISDN (Q.931) and offers enhanced PBX features in a private network. (→ 1.27.3 QSIG Network)

Voice over Internet Protocol (VoIP) Network

The PBX can connect to another PBX via an IP-type private network. In this case, voice signals are converted into IP packets and sent through this network. (→ 1.27.4 Voice over Internet Protocol (VoIP) Network)

Built-in Small Call Centre Features

An incoming call distribution group (→ 1.2.2 Incoming Call Distribution Group Features) can be used as a small call centre with the following features:

Queuing Feature

When a preprogrammed number of extensions in an incoming call distribution group are busy, additional incoming calls can wait in a queue. While calls are waiting in the queue, the calls are handled by the Queuing Time Table, which can be assigned for each time mode (day/lunch/break/night). (→ 1.2.2.3 Queuing Feature)

Log-in/Log-out

Incoming call distribution group members can join (**Log-in**) or leave (**Log-out**) the groups manually. While logged-in, a member extension can have a preprogrammed time period automatically for refusing calls after completing the last call (**Wrap-up**). (→ 1.2.2.6 Log-in/Log-out)

VIP Call

It is possible to assign a priority to incoming call distribution groups. If an extension belongs to multiple groups and the extension becomes idle, queuing calls in the groups will be distributed to the extension in priority order. (→ 1.2.2.4 VIP Call)

Computer Telephony Integration (CTI) Features

Connecting a personal computer (PC) to this PBX (via a DPT, or via a Server PC on a LAN) enables extension users to make use of advanced features by using the stored data in the PC or in the Server PC. (→ 1.28.1 Computer Telephony Integration (CTI))

Voice Mail Features

This PBX supports Voice Processing Systems (VPS) with DTMF Integration as well as DPT (Digital) Integration. (→ 1.22 Voice Mail Features)

Portable Station (PS) Features

PSs (e.g., KX-TD7590, KX-TD7690) can be connected to this PBX. It is possible to use the PBX features using the PS like a PT. A PS can also be used in parallel with a wired telephone (**Wireless XDP Parallel Mode**). In this case, the wired telephone is the main telephone and the PS is the sub telephone. (→ 1.23 Portable Station (PS) Features)

PC Phone/PC Console Features

This PBX supports PC Phone and PC Console. These Panasonic CTI applications provide advanced features.

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Section 1

Call Handling Features

1.1 Incoming Call Features

1.1.1 Incoming Trunk Call Features

1.1.1.1 Incoming Trunk Call Features—SUMMARY

Description

Incoming calls via a trunk (public line) are distributed to their destination using a suitable distribution feature.

1. Available Networking Type for Each Optional Trunk Card Type

Each trunk port of an optional trunk card can be assigned its networking type: Public, Private, or VPN (Virtual Private Network).

Trunk Card Type / Networking Type / Channel Type		Public (DIL/DID/ DDI/MSN)	Private (TIE)*1	Virtual Private Network (VPN)*2
LCOT	—	✔*		
DID	—	✔*		
T1	LCOT	✔*		
	GCOT	✔*		
	DID	✔*		
	TIE (E & M)	✔	✔*	
	OPX (EXTN.)			
E1	DR2	✔*		
	E & M-C	✔	✔*	
	E & M-P	✔	✔*	
E&M	—	✔	✔*	
BRI/PRI	CO	✔*		✔
	EXTN.			
	QSIG-Master		✔*	
	QSIG-Slave		✔*	
IP-GW	—		✔*	

Note: ✔*: Enable (default), ✔: Enable

*1: → 1.27.1 TIE Line Service

*2: → 1.27.2 Virtual Private Network (VPN)

2. Distribution Feature

One of the following features can be assigned to each trunk port:

Feature	Description	Details in
Direct In Line (DIL)	Directs a call to a preprogrammed single destination (e.g., Operator).	• 1.1.1.2 Direct In Line (DIL)
Direct Inward Dialling (DID)	Directs a call with a DID number from a DID line to a preprogrammed destination. DID is also known as Direct Dialling In (DDI).	• 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)
Multiple Subscriber Number (MSN) Ringing Service	Directs a call with an MSN from an ISDN line to a preprogrammed destination.	• 1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service

3. Destination Change with the Caller's Identification Number

The Calling Line Identification (CLI) Distribution feature works in conjunction with the DIL/DID/DDI/MSN features.

Feature	Description	Details in
Calling Line Identification (CLI) Distribution	Directs a call to a CLI destination if the caller's identification number has been assigned to the Caller ID Table.	• 1.1.1.5 Calling Line Identification (CLI) Distribution

4. Available Distribution Feature for Each Optional Trunk Card Type

Trunk Card Type / Feature: Channel Type		DIL	DID/DDI	MSN
LCOT	—	✔*		
DID	—	✔	✔*	
T1	LCOT	✔*		
	GCOT	✔*		
	DID	✔	✔*	
	TIE (E & M)	✔*	✔	
E1	DR2	✔	✔*	
	E & M-C	✔*	✔	
	E & M-P	✔*	✔	
E&M	—	✔*		
BRI	CO	✔	✔*	✔
PRI	CO	✔	✔*	

Note: ✔*: Enable (default), ✔: Enable

5. Available Destination

Destination	Availability
Wired Extension (PT/SLT/ISDN Extension/T1-OPX)	✔
PS	✔
Incoming Call Distribution Group	✔
PS Ring Group	✔
VM Group (DTMF/DPT)	✔
External Pager (TAFAS)	✔
DISA	✔
Analogue/ISDN Remote Maintenance	✔
Idle Line Access no. + Phone no.	
Trunk Group Access no. + Trunk Group no. + Phone no.	
Other PBX Extension (TIE with no PBX Code)	✔
Other PBX Extension (TIE with PBX Code)	

6. Intercept Routing

After distribution, the following features may be required.

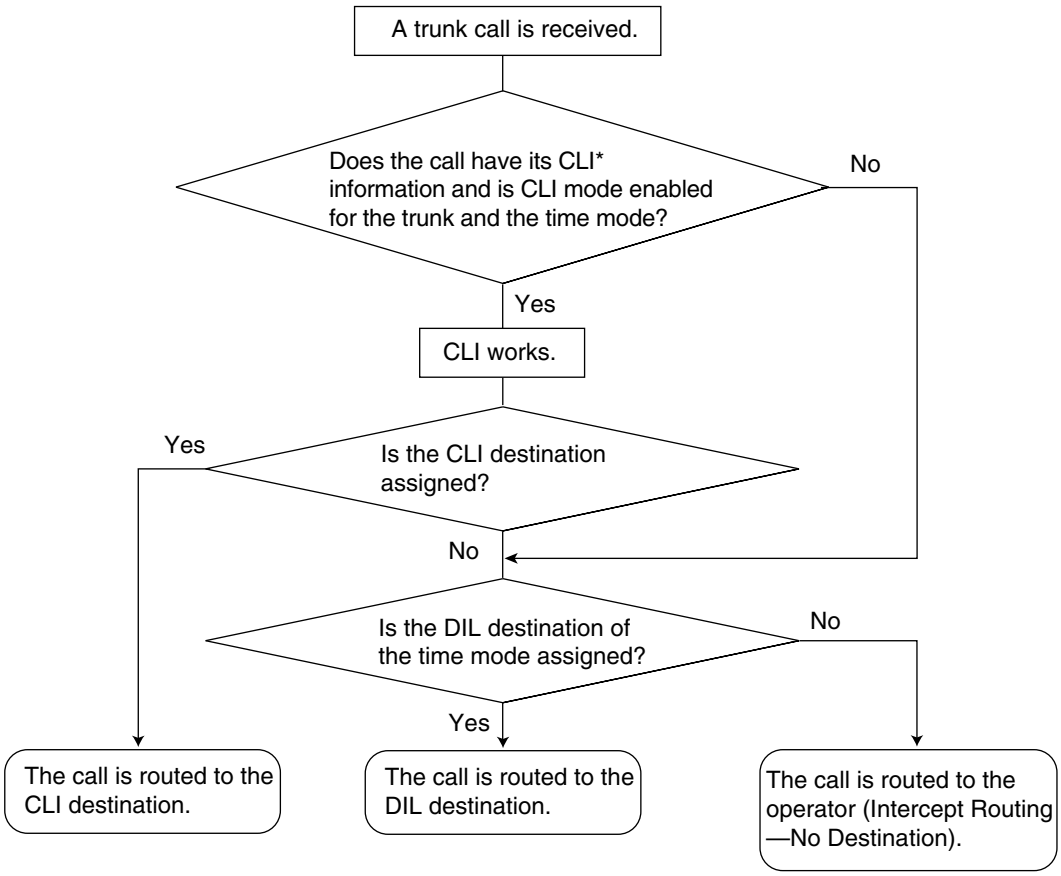
Feature		Description	Details in
Intercept Routing	**No Answer (IRNA)**	If a called party does not answer a call within a preprogrammed time period (Intercept time), it is redirected to the preprogrammed destination.	• 1.1.1.6 Intercept Routing
	Busy/DND	If a called party is busy or in DND mode, the call is redirected to the preprogrammed destination.	
	No Destination	If a destination is not assigned, the call is redirected to the operator.	• 1.1.1.7 Intercept Routing—No Destination

1.1.1.2 Direct In Line (DIL)

Description

Provides automatic direction of an incoming trunk call to a preprogrammed destination. Each trunk has a destination for each time mode (day/lunch/break/night).

[Method Flowchart]


A trunk call is received.
Does the call have its CLI* information and is CLI mode enabled for the trunk and the time mode?
No
Yes
CLI works.
Yes
Is the CLI destination assigned?
No
Is the DIL destination of the time mode assigned?
No
Yes
The call is routed to the CLI destination.
The call is routed to the DIL destination.
The call is routed to the operator (Intercept Routing —No Destination).

*: Calling Line Identification (CLI) Distribution:
If the CLI routing is enabled and the caller's identification number is assigned in the Caller ID Table, the call will not be routed to the DIL destination, but routed to the CLI destination.

[Programming Example of DIL Table]

The table can be programmed for each trunk.

Trunk No.	CLI				Destination*			
	Day	Lunch	Break	Night	Day	Lunch	Break	Night
01	Enable	Disable	Enable	Disable	101	100	101	100
02	Enable	Disable	Disable	Disable	102	100	102	100
:	:	:	:	:	:	:	:	:

*: → DIL 1:1 Destination [450]

Note

Tenant number and VPS trunk group number can also be assigned in the DIL table. Tenant number is used to determine the time mode (day/lunch/break/night) for the corresponding trunk. VPS trunk group number is used in Voice Mail DPT (Digital) Integration (→ 1.22.3 Voice Mail DPT (Digital) Integration).

Explanation:

If a trunk call is received from trunk 01;
In Day mode: CLI is enabled. Route to CLI destination.
In Lunch mode: CLI is disabled. Route to DIL destination, extension 100.

Feature Guide References

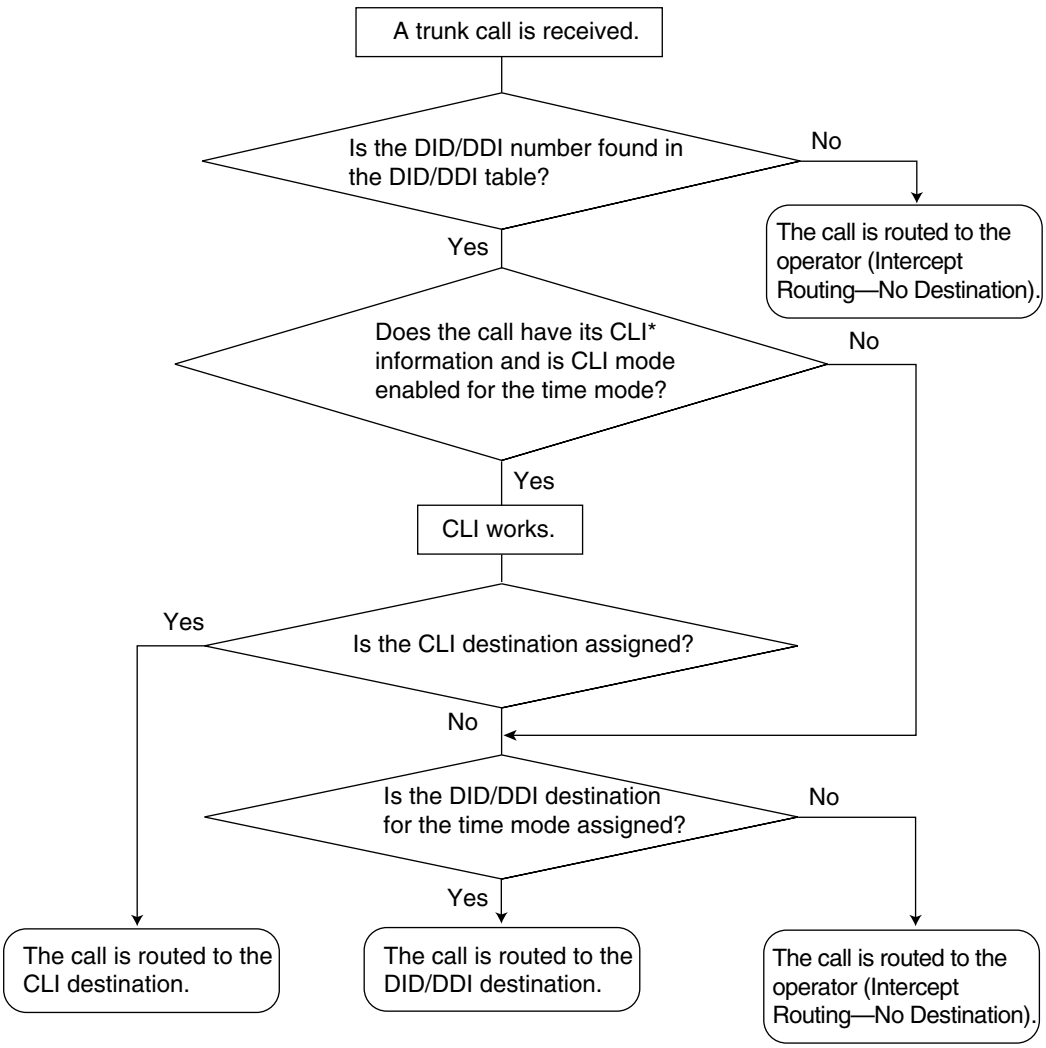
1.1.1.5 Calling Line Identification (CLI) Distribution
2.2.4 Time Service

1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)

Description

Provides automatic direction of an incoming call with a DID/DDI number to a preprogrammed destination. Each DID/DDI number has a destination for each time mode (day/lunch/break/night).

[Method Flowchart]


A trunk call is received.
Is the DID/DDI number found in the DID/DDI table?
No
The call is routed to the operator (Intercept Routing—No Destination).
Yes
Does the call have its CLI* information and is CLI mode enabled for the time mode?
No
Yes
CLI works.
Yes
Is the CLI destination assigned?
No
Is the DID/DDI destination for the time mode assigned?
No
Yes
The call is routed to the CLI destination.
The call is routed to the DID/DDI destination.
The call is routed to the operator (Intercept Routing—No Destination).

*: Calling Line Identification (CLI) Distribution:
If the CLI routing is enabled and the caller's identification number is assigned in the Caller ID Table, the call will not be routed to the DID/DDI destination, but routed to the CLI destination.

[Programming Example of DID/DDI Table]

DDI can be programmed as DID.

Location	DID/DDI No.*1	DID/DDI Name*2	CLI				DID/DDI Destination*3			
			Day	Lunch	Break	Night	Day	Lunch	Break	Night
0001	123-4567	John White	Enable	Disable	Enable	Disable	105	100	105	100
0002	123-2468	Tom Smith	Enable	Disable	Disable	Disable	102	100	102	100
0003	123-456	A company	Enable	Disable	Disable	Disable	101	101	101	100
:	:	:	:	:	:	:	:	:	:	:

*1: → DID Number [451]
*2: → DID Name [452]
*3: → DID Destination [453]

Note

Tenant number and VPS trunk group number can also be assigned in the DID/DDI table. Tenant number is used to determine the time mode (day/lunch/break/night) for the corresponding DID/DDI number. VPS trunk group number is used in Voice Mail DPT (Digital) Integration (→ 1.22.3 Voice Mail DPT (Digital) Integration).

Explanation:

If the DID/DDI number is "123-4567":

1. Checks the number in the table.

→ Matches the number in location 0001.

2. Checks the time mode.

In Day mode: CLI is enabled. Route to CLI destination.
In Lunch mode: CLI is disabled. Route to DID/DDI destination, extension 100.

Conditions

- To use this feature, the DID/DDI service must be assigned for a trunk port as the distribution method (→ BRI DIL/DDI/MSN Selection [421] *For BRI only).
- **DID/DDI Number Modification**

  It is possible to modify a received DID/DDI number, which may be convenient when programming the DID/DDI table. Modification method (removed number of digits/added number) can be programmed on a trunk port basis.

  [Modification Example]

  Removed number of digits: 6
  Added number: 10
  Received DID/DDI number: 87654321 ➡

  Modified DID/DDI number: ~~876543~~ 21 = 1021

  1) Remove the first 6 digits.

  2) Add "10".

- **The Inter-digit Time**
  When the Inter-digit time expires, the PBX stops receiving DID/DDI number and starts to check the DID/DDI table. (Refer to the [Programming Example of DID/DDI Table] above). Even if the Inter-digit time does not expire, the PBX stops receiving the DID/DDI number when the received number is found in the DID/DDI table. The PBX then routes the call to the corresponding destination. If the received number matches several DID/DDI numbers in the table, the DID/DDI number of the lowest numbered location has priority.
  [Example] If a call is received in Lunch mode;

Received Number	Destination	Explanation
123-4567	Extn. 100	The PBX finds the match in location 0001 in the table after receiving "7". So the call is routed to extension 100.
123-456	Extn. 101	The Inter-digit time expired after receiving "6". The PBX finds the match in location 0003 in the table. So the call is routed to extension 101.

Feature Guide References

1.1.1.5 Calling Line Identification (CLI) Distribution

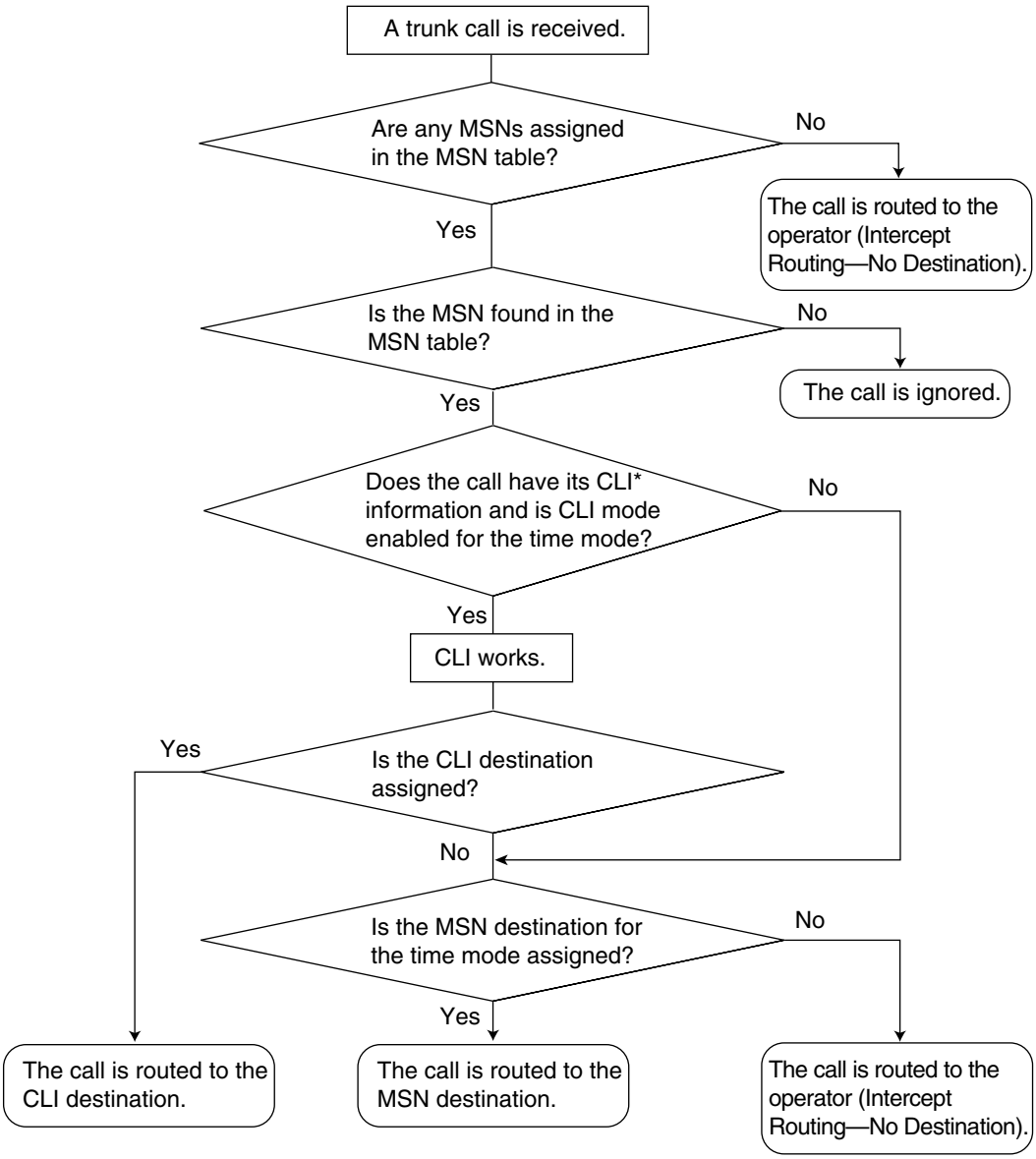
2.2.4 Time Service

1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service

Description

Provides automatic direction of an incoming ISDN-BRI (Basic Rate Interface) line call with an MSN to a preprogrammed destination. One ISDN-BRI port can support a maximum of 10 MSNs. Each MSN has a destination for each time mode (day/lunch/break/night).
Point-to-multipoint must be selected for the ISDN configuration (→ BRI Configuration [426]).

[Method Flowchart]


A trunk call is received.
Are any MSNs assigned in the MSN table?
No
The call is routed to the operator (Intercept Routing—No Destination).
Yes
Is the MSN found in the MSN table?
No
The call is ignored.
Yes
Does the call have its CLI* information and is CLI mode enabled for the time mode?
No
Yes
CLI works.
Yes
Is the CLI destination assigned?
No
Is the MSN destination for the time mode assigned?
No
Yes
The call is routed to the CLI destination.
The call is routed to the MSN destination.
The call is routed to the operator (Intercept Routing—No Destination).

*: Calling Line Identification (CLI) Distribution:
If the CLI routing is enabled and the caller's identification number is assigned in the Caller ID Table, the call will not be routed to the MSN destination, but routed to the CLI destination.

[Programming Example of MSN Table for ISDN BRI Port 1]
A table can be programmed for each ISDN-BRI port. Each BRI port has 10 MSN locations.

Location	MSN	MSN Name	CLI				MSN Destination			
			Day	Lunch	Break	Night	Day	Lunch	Break	Night
01	123-4567	A Company	Enable	Disable	Enable	Disable	101	100	101	100
02	123-2468	C Company	Enable	Disable	Disable	Disable	102	100	102	100
:	:	:	:	:	:	:	:	:	:	:
10	:	:	:	:	:	:	:	:	:	:

Note

Tenant number and VPS trunk group number can also be assigned in the MSN table. Tenant number is used to determine the time mode (day/lunch/break/night) for the corresponding MSN. VPS trunk group number is used in Voice Mail DPT (Digital) Integration (→ 1.22.3 Voice Mail DPT (Digital) Integration).

Explanation:

If the MSN "123-4567" is received from BRI port 1:

1. Checks the number in the table.

→ Matches the number in location 01.

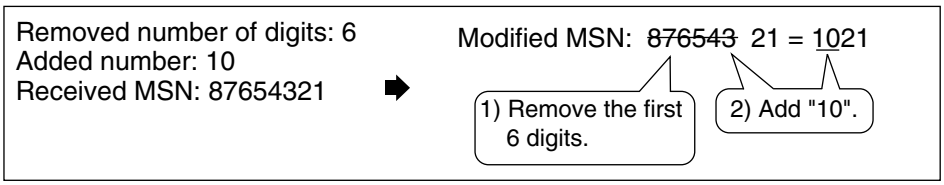
2. Checks the time mode.

In Day mode: CLI is enabled. Route to CLI destination.
In Lunch mode: CLI is disabled. Route to MSN destination, extension 100.

Conditions

- To use this feature, the MSN service must be assigned for a trunk port as the distribution method (→ BRI DIL/DDI/MSN Selection [421]).
- **MSN Modification**
  It is possible to modify a received MSN to make it shorter, which may be convenient when programming the MSN table. Modification method (removed number of digits/added number) can be programmed on a trunk port basis.

  [Modification Example]


Removed number of digits: 6
Added number: 10
Received MSN: 87654321
Modified MSN: 876543 21 = 1021
1) Remove the first 6 digits.
2) Add "10".

- When using point-to-multipoint configuration with a BRI, do not connect another ISDN terminal device in parallel with the PBX. As only two channels can be used at one time with the BRI, the other ISDN terminal device may monopolise both channels.

Feature Guide References

1.1.1.5 Calling Line Identification (CLI) Distribution

2.2.4 Time Service

1.1.1.5 Calling Line Identification (CLI) Distribution

Description

Directs an incoming trunk call to a destination when the caller's identification number (e.g., Caller ID) matches the number in the System Speed Dialling Table which is used as the Caller ID Table. Each Caller ID number (telephone number for each System Speed Dialling number) can have its own destination.

CLI Feature	Description	Details in
Caller ID	Caller's number is sent from an analogue trunk.	• 1.16.1 Caller ID
Calling Line Identification Presentation (CLIP)	Caller's number is sent from an ISDN line.	• 1.19.1.2 Calling/ Connected Line Identification Presentation (CLIP/COLP)
Automatic Number Identification (ANI) (KX-TDA100/KX-TDA200 only)	Caller's number is sent from an E1 or T1 line.	• 1.20.1 E1 Line Service (KX-TDA100/ KX-TDA200 only) • 1.21.1 T1 Line Service (KX-TDA100/ KX-TDA200 only)

CLI always works in conjunction with the following call distribution methods:

a) DIL

b) DID/DDI

c) MSN Ringing Service

Each trunk (for DIL) and the DID/DDI/MSN number can enable or disable the CLI feature for each time mode (day/lunch/break/night) (→ 2.2.4 Time Service).
When the call has its Caller ID and the CLI is enabled for the time mode, the call will be handled by the CLI method.

[Programming Example of System Speed Dialling Table for CLI]

Location (System Speed Dialling No.)	Telephone No.*1	System Speed Dialling Name*2	CLI Destination
000	901234567890	ABC Company	200
001	:	:	:
:	:	:	:

*1: → System Speed Dialling Number [001]
*2: → System Speed Dialling Name [002]

Explanation:

If the caller's number is "0123-456-7890" (The Trunk Access number is disregarded.):

1. Checks the number in the table.

→ Matches the number in location 000.

2. The call is routed to the CLI destination, extension 200.

Conditions

- **Automatic Caller ID Number Modification**
  The Caller ID number is used after modification by the Automatic Caller ID Number Modification. (→ 1.16.1 Caller ID)

Feature Guide References

1.1.1.2 Direct In Line (DIL)

1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)

1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service

1.6.1.5 Speed Dialling—Personal/System

1.1.1.6 Intercept Routing

Description

Provides automatic redirection of incoming trunk calls. There are two types of Intercept Routing as follows:

Feature	Description
Intercept Routing—No Answer (IRNA)	If a called party does not answer a call within a preprogrammed time period (Intercept time) (→ Intercept Time [203]), it is redirected to the preprogrammed destination.
Intercept Routing—Busy/DND	If a called party is busy or in DND mode, the call is redirected to the preprogrammed destination.

The available intercept destination is as follows:

Type 1: The destination assigned on the extension port which the original destination joins. (→ Extension Intercept Destination [604])

Type 2: The destination assigned on the trunk group which receives the call. (→ Trunk Group Intercept Destination [470])

Original Destination	The Available Intercept Destination
Wired Extension (PT/SLT/ISDN Extension/T1-OPX)	Type 1
PS	Type 1
Incoming Call Distribution Group	Intercept Routing—Overflow in an Incoming Call Distribution Group works (→ 1.2.2.5 Overflow Feature). The overflow destination is assigned on the incoming call distribution group which the original destination joins (→ Destination for Overflow Time Expiration [625]).
PS Ring Group	Type 2
VM Group (DTMF/DPT)	Type 2
External Pager (TAFAS)	Type 2
DISA	Type 2*
Analogue/ISDN Remote Maintenance	Not available
Idle Line Access no. + Phone no.	Not available
Trunk Group Access no. + Trunk Group no. + Phone no.	Not available
Other PBX Extension (TIE with no PBX Code)	Not available
Other PBX Extension (TIE with PBX Code)	Not available

*: This is applied only when a trunk call arrives on a DISA line but the line is busy at that time. Once the call reaches the destination extension by using the DISA feature, the Intercept Routing feature of the extension works.

Each of them can have different intercept destinations for each time mode (day/lunch/break/night) (→ 2.2.4 Time Service).

[Available Intercept Destination]

Intercept Destination	Availability
Wired Extension (PT/SLT/ISDN Extension/T1-OPX)	✔
PS	✔
Incoming Call Distribution Group	✔
PS Ring Group	✔
VM Group (DTMF/DPT)	✔
External Pager (TAFAS)	✔
DISA	✔
Analogue/ISDN Remote Maintenance	✔
Idle Line Access no. + Phone no.	✔
Trunk Group Access no. + Trunk Group no. + Phone no.	✔
Other PBX Extension (TIE with no PBX Code)	✔
Other PBX Extension (TIE with PBX Code)	✔

Conditions

- **Intercept Routing—Busy/DND on/off**
  Each Intercept Routing—Busy and Intercept Routing—DND can be enabled or disabled through system programming.
  If disabled, one of the following is activated depending on the trunk card type which a call arrives through:
  - **a) LCOT or T1 (LCOT/GCOT) Card:** The incoming trunk call will ring at the original destination while the caller hears a ringback tone.
  - **b) Other Trunk Cards:** Busy tone will be sent to the caller.
- **If the intercept destination cannot receive the call:**
  - **a) Intercept Routing—No Answer:** Intercept timer will restart at the original destination, until the call is answered.
  - **b) Intercept Routing—Busy/DND:** The call will be sent back to the original destination when the call arrives through the LCOT or T1 (LCOT/GCOT) card. When the call arrives through other trunk cards the caller will hear busy tone.
- **Idle Extension Hunting**
  If an extension is a member of an idle extension hunting group, calls to that extension will not be redirected by Intercept Routing—Busy/DND. If the extension is busy or in DND mode, calls to that extension will be redirected to the next extension in the idle extension hunting group.

Feature Guide References

1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)

1.1.1.7 Intercept Routing—No Destination

Description

Provides automatic redirection of an incoming trunk call which does not have a destination assigned. The intercept destination is an operator (tenant/PBX) (→ Operator Assignment [006] *For PBX operator only).

Conditions

- **Intercept Routing—No Destination on/off**
  The Intercept Routing—No Destination feature can be enabled or disabled through system programming.
  If disabled, reorder tone will be sent to the caller. However, the Intercept Routing—No Destination feature always works for calls through the LCOT or T1 (LCOT/GCOT) card even when disabled.
- **If an operator (tenant/PBX) is not assigned:**
  The lowest jack numbered extension will be the intercept destination.
- **Intercept Routing—No Destination also applies to:**
  Calls from doorphones.

Feature Guide References

2.2.5 Operator Features

1.1.2 Internal Call Features

1.1.2.1 Internal Call Features—SUMMARY

Description

The following incoming calls reach their destination:

Feature	Description	Details in
Intercom Call	A call from one extension to another.	• 1.5.3 Intercom Call
Doorphone Call	When a call from a doorphone reaches its destination, the recipient can talk to the visitor.	• 1.15.1 Doorphone Call

[Available Destination]

The destinations of doorphone calls can be assigned for each time mode (day/lunch/break/night) (→ 2.2.4 Time Service) on a doorphone port basis (→ Doorphone Call Destination [720]).

Destination	Calling from	
	Extension	Doorphone
Wired Extension (PT/SLT/ISDN Extension/T1-OPX)	✔	✔
PS	✔	✔
Incoming Call Distribution Group	✔	✔
PS Ring Group	✔	✔
VM Group (DTMF/DPT)	✔	✔
External Pager (TAFAS)	✔	✔
DISA		
Analogue/ISDN Remote Maintenance	✔	
Idle Line Access no. + Phone no.	✔	✔
Trunk Group Access no. + Trunk Group no. + Phone no.	✔	✔
Other PBX Extension (TIE with no PBX Code)	✔	✔
Other PBX Extension (TIE with PBX Code)	✔	✔

✔: Available

1.1.2.2 Internal Call Block

Description

Internal calls can be restricted on a COS basis. Each COS of the caller can enable or disable this feature for each COS of the called party.

[Programming Example]

Caller	Called Party			
	COS 1	COS 2	COS 3	...
COS 1				
COS 2		✔	✔	✔
COS 3	✔	✔		✔
:	:	:	:	:

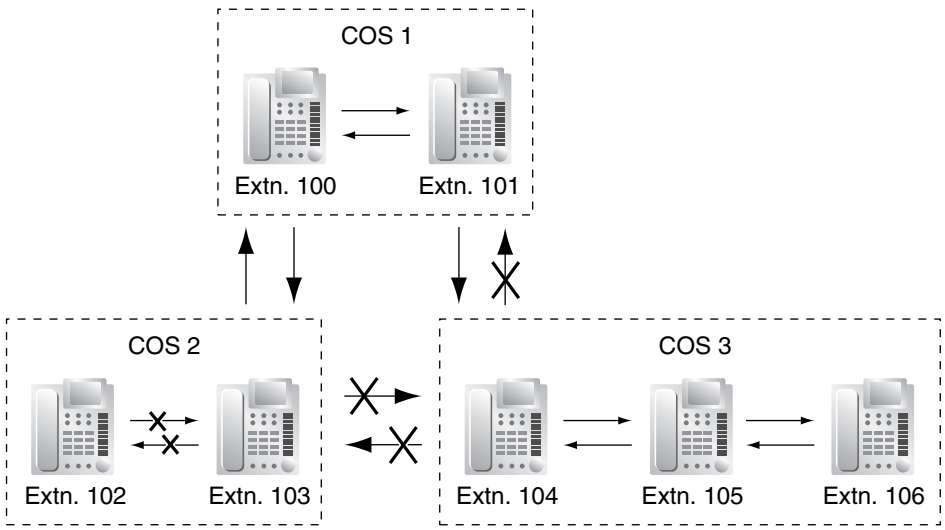
✔: Block

Explanation:

a) COS 1 can make calls to all extensions.

b) COS 2 can make calls to COS 1 only. (COS 2 cannot make calls to COS 2.)

c) COS 3 can make calls to COS 3 only.



Conditions

- Restricted extension numbers cannot be the parameter of the feature settings (e.g., Walking Extension).
- All extensions can make an Operator Call (→ 2.2.5 Operator Features) regardless of the Internal Call Block.

- This feature can also restrict calling a doorphone from an extension on a COS of the extension and a COS of the doorphone port basis. Each doorphone port can be assigned a COS. (→ 1.15.1 Doorphone Call)

1.1.3 Incoming Call Indication Features

1.1.3.1 Incoming Call Indication Features—SUMMARY

Description

Incoming calls are indicated by various methods as follows:

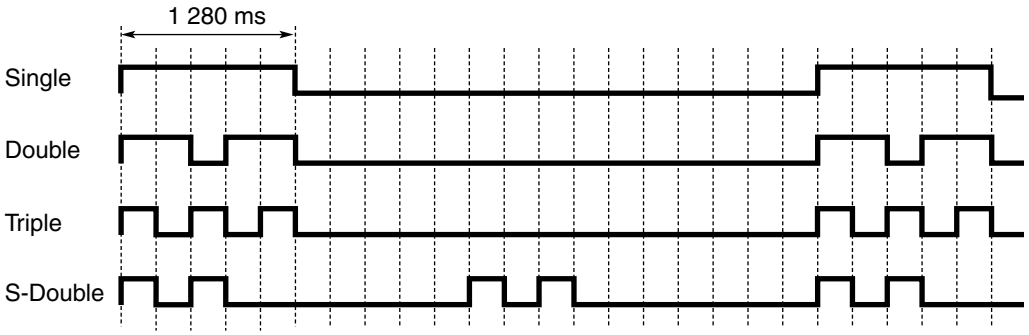
Type	Feature	Description	Details in
Ring Tone	**Ring Tone Pattern Selection**	A telephone rings when receiving a call. The ring tone patterns can be changed for each incoming call type.	• 1.1.3.2 Ring Tone Pattern Selection
Voice-calling	**Alternate Receiving—Ring/Voice**	A PT user can select to receive intercom calls by ring tone or by voice, through personal programming.	• 1.5.3 Intercom Call
LED (LED: Light Emitting Diode)	**LED Indication**	The light shows line conditions with a variety of light patterns.	• 1.18.3 LED Indication
Display (Caller's Information)	**Display Information**	The display shows the caller's information.	• 1.18.4 Display Information
External Pager	**Trunk Answer from Any Station (TAFAS)**	The external pager sends a ring tone when receiving a call.	• 1.15.3 Trunk Answer From Any Station (TAFAS)
Tone/Voice during a Conversation	**Call Waiting**	A busy extension hears a tone, or voice from the handset/built-in speaker indicating that another incoming call is waiting.	• 1.1.3.3 Call Waiting

1.1.3.2 Ring Tone Pattern Selection

Description

A ring tone pattern can be selected for each incoming call type on a Ring Tone Pattern Table basis, which can be assigned for each extension.

[Ring Tone Patterns]



[Programming Example of Ring Tone Pattern Table]

There are a specified number of programmable tables each of which allows the assignment of ring tone patterns for the following incoming call types:
For trunk calls (including trunk call hold recalls) and doorphone calls, select ring tone patterns on a trunk group or a doorphone port basis.

Table No.	Intercom Call/Hold Recall	Trunk Call/Hold Recall			Doorphone Call			Timed Reminder	Call Back	LCS
		TRG1	TRG2	...	Port 1	Port 2	...			
1	Double	Single			Single					
2	Single	Double			Double					
:	:	:	:	:	:	:	:	:	:	:

Each extension can select one of the tables.

Conditions

- "PT Ring off Setting" can be enabled or disabled through system programming. If disabled, PT users cannot set the ring off.

1.1.3.3 Call Waiting

Description

Used to inform a busy extension that another incoming call is waiting. The busy extension user can answer the second call by disconnecting the current call or placing it on hold.
The following notification method can be assigned for each extension depending on the call waiting and the telephone type:

a) Call Waiting Tone: Tone from the handset or built-in speaker

b) OHCA: Voice from the built-in speaker

c) Whisper OHCA: Voice from the handset

d) Off: No notification.

Call Type	Notification Method	
	DPT	Other Telephone
Intercom Call	Call Waiting tone/OHCA/ Whisper OHCA/Off	Call Waiting tone/Off
Trunk Call*	Call Waiting tone/Off	

*: Including a doorphone call, call via an incoming call distribution group, and a transferred trunk call from another extension.

This feature is also known as Busy Station Signalling (BSS).

Conditions

- Call Waiting call for an extension in a VM group (DPT/DTMF) is not available.
- **Data Line Security**
  Setting Data Line Security cancels the Call Waiting setting. (→ 1.10.5 Data Line Security)
- **Call Waiting Tone**
  A PT user can hear different Call Waiting tones for trunk call and intercom call if "Tone 2" has been selected through personal programming (Call Waiting Tone Type Selection). If "Tone 1" has been selected, the same Call Waiting tone will be heard for both trunk call and intercom call.
  All Call Waiting tone patterns have a default (→ 4.2.1 Tones/Ring Tones).
- **Caller Information**
  With the Call Waiting tone, the caller's information flashes on the display for five seconds at 15-second intervals.
- **Call Waiting from the Telephone Company**
  Besides the Call Waiting service within the PBX, the Call Waiting tone offered by an analogue line from your telephone company informs the extension user of another incoming trunk call that is waiting. He can answer the second call by disconnecting the current call or placing it on hold. For details, consult your telephone company.
  Call Waiting Caller ID (Visual Caller ID):
  With the Call Waiting tone offered by an analogue line from the telephone company, the Caller ID number can be received and flash on the display twice for five seconds at 15-second intervals.

Feature Guide References

1.7.4 Second Call Notification to Busy Extension

User Manual References

User Manual

1.4.4 Answering a Call Waiting

1.7.3 Receiving Call Waiting (Call Waiting/Off-hook Call Announcement [OHCA]/Whisper OHCA)

3.1.2 Settings on the Programming Mode

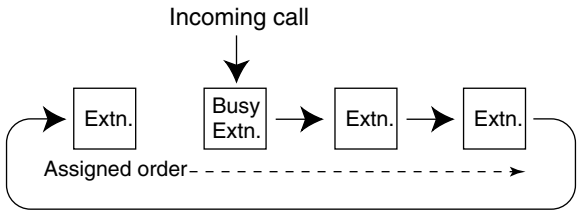
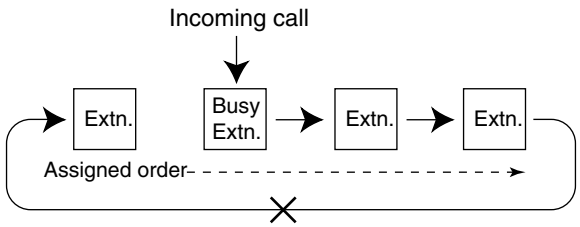
1.2 Receiving Group Features

1.2.1 Idle Extension Hunting

Description

If a called extension is busy or in DND mode, Idle Extension Hunting redirects the incoming call to an idle member of the same idle extension hunting group, which can be programmed through system programming (→ Idle Extension Hunting Group Member [681]). Idle extensions are automatically searched according to a preprogrammed hunting type (→ Idle Extension Hunting Type [680]).

This feature is also known as Station Hunting.

Type	Description
Circular Hunting	An idle extension is searched for in the order specified in the idle extension hunting group in a circular way. 
Terminated Hunting	An idle extension is searched for in the order specified in the idle extension hunting group until reaching the last assigned extension. 

Conditions

- **Idle Extension Hunting applies to:**
  Intercom, trunk, and doorphone calls to a single destination.
- An extension user can belong to only one idle extension hunting group.

- **If all the searched extensions are busy:**
  The PBX redirects the call to an overflow destination which can be assigned for each idle extension hunting group and each time mode (day/lunch/break/night) (→ 2.2.4 Time Service).
  [Available Destination]

Destination	Availability
Wired Extension (PT/SLT/ISDN Extension/T1-OPX)	✔
PS	✔
Incoming Call Distribution Group	✔
PS Ring Group	✔
VM Group (DTMF/DPT)	✔
External Pager (TAFAS)	✔
DISA	✔
Analogue/ISDN Remote Maintenance	✔
Idle Line Access no. + Phone no.	✔
Trunk Group Access no. + Trunk Group no. + Phone no.	✔
Other PBX Extension (TIE with no PBX Code)	✔
Other PBX Extension (TIE with PBX Code)	✔

- **FWD/DND Mode**
  While searching for an idle extension within an idle extension hunting group, any extension which has set FWD—All Calls or DND feature will be skipped, and the call will go to the next extension in the group.

Feature Guide References

1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)

1.2.2 Incoming Call Distribution Group Features

1.2.2.1 Incoming Call Distribution Group Features—SUMMARY

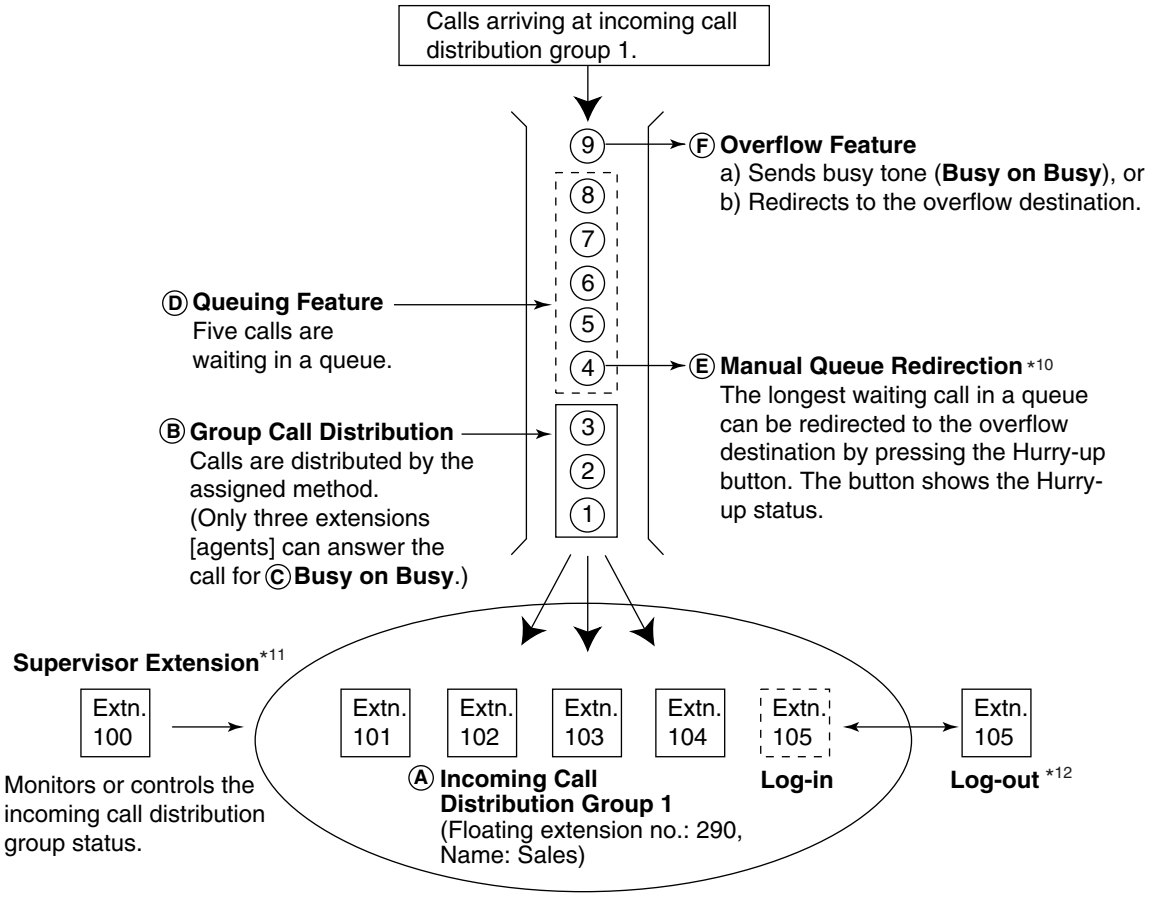
Description

An incoming call distribution group is a group of extensions programmed through system programming (→ Incoming Call Distribution Group Member [620]). An incoming call distribution group receives calls directed to the group. Each incoming call distribution group has a floating extension number.
Incoming calls directed to an incoming call distribution group are distributed to the member extensions in the group using a distribution method. When a preprogrammed number of extensions in the group are busy, the incoming calls can wait in a queue.
Each incoming call distribution group and member extensions can be programmed as desired to handle incoming calls. Calls to the group can be monitored by an extension assigned as a supervisor (supervisor extension).

Programming Item Example for Incoming Call Distribution Group 1 with Diagram
A through F in the table are described in the following diagram.

A			B	C	D	E	F					
Group No.	Floating Extn. No.*1	Group Name*2	Distribution Method*3	Max. No. of Busy Extensions*4	Queuing Call Capacity*5	Hurry-up Level*6	Overflow Time*7	Overflow Destination*8				Tenant No.*9
								Day	···	···	Night	
1	290	Sales	Ring	3	5	3	60	100	···	···	100	1
2	291	Engineering	UCD	Max.	11	8	90	200	···	···	200	5
3												
:												

*1: → Incoming Call Distribution Group Floating Extension Number [622]
*2: → Incoming Call Distribution Group Name [623]
*3: → Incoming Call Distribution Group Distribution Method [624]
*4: → Maximum Number of Agents [632]
*5: → Queuing Call Capacity [628]
*6: → Queuing Hurry-up Level [629]
*7: → Overflow Time [626]
*8: → Destination for Overflow Time Expiration [625]/Destination When All Busy [627]
*9: Tenant number is required to determine the time mode (day/lunch/break/night) (→ 2.2.4 Time Service) and the music source (for Music on Hold) for each group.


Calls arriving at incoming call distribution group 1.
9
8
7
6
5
4
3
2
1
Ⓕ Overflow Feature
a) Sends busy tone (Busy on Busy), or
b) Redirects to the overflow destination.
Ⓓ Queuing Feature
Five calls are waiting in a queue.
Ⓔ Manual Queue Redirection *10
The longest waiting call in a queue can be redirected to the overflow destination by pressing the Hurry-up button. The button shows the Hurry-up status.
Ⓑ Group Call Distribution
Calls are distributed by the assigned method.
(Only three extensions [agents] can answer the call for Ⓒ Busy on Busy.)
Supervisor Extension*11
Extn. 100
Monitors or controls the incoming call distribution group status.
Extn. 101
Extn. 102
Extn. 103
Extn. 104
Extn. 105
Ⓐ Incoming Call Distribution Group 1
(Floating extension no.: 290, Name: Sales)
Log-in
Extn. 105
Log-out *12

*10: → 1.2.2.3 Queuing Feature
*11: → 1.2.2.7 Supervisory Feature
*12: → 1.2.2.6 Log-in/Log-out

1. **Group Call Distribution** [→ 1.2.2.2 Group Call Distribution]
   Incoming calls are distributed using one of the following methods:

Distribution Method	Description
Uniform Call Distribution (UCD)	Calls go to a different extension uniformly each time a call is received.
Priority Hunting	An idle extension is searched in the specified order.
Ring	All extensions in the incoming call distribution group ring simultaneously.

2. **Queuing Feature** [→ 1.2.2.3 Queuing Feature]
   If a preprogrammed numbers of extensions in an incoming call distribution group are busy, a preprogrammed number of additional calls can wait in a queue.
   While calls are waiting in the queue, an outgoing message (OGM) or Music on Hold can be sent to the waiting callers.

3. **VIP Call** [→ 1.2.2.4 VIP Call]
   It is possible to assign a priority to incoming call distribution groups so that an incoming

call can be received from the groups in priority order.

4. **Overflow Feature** [→ 1.2.2.5 Overflow Feature]
   A call is redirected to a preprogrammed destination when it cannot be answered or queued (**Intercept Routing—Overflow in an Incoming Call Distribution Group**). It is also possible to send busy tone (**Busy on Busy**) or disconnect the line.

5. **Incoming Call Distribution Group Controlling Feature**

Feature		Description	Details in
Log-in/Log-out		Member extensions can join the group to handle calls (Log-in) or leave the group for a break (Log-out). They can leave the group temporarily when they are away from their desks, to prevent calls being sent to their extensions.	• 1.2.2.6 Log-in/Log-out
Supervisory Feature	**Incoming Call Queue Monitor**	The supervisor extension can monitor various information about the incoming calls for each incoming call distribution group on his display.	• 1.2.2.7 Supervisory Feature
	Log-in/Log-out Monitor and Remote Control	**Monitor:** The supervisor extension can monitor the log-in/log-out status of the group members. **Remote Control:** The supervisor extension can change the status of the members.	

Conditions

- One extension can belong to multiple incoming call distribution groups.
- **ICD Group button**
  An Incoming Call Distribution (ICD) Group button can be assigned on a flexible button for each incoming call distribution group. It receives the incoming calls to the group.
  One extension can have more than one ICD Group button of the same or different incoming call distribution groups **(Multiple ICD Group)**. If all ICD Group buttons in the same incoming call distribution group are occupied, the next incoming call will be held in a queue or will overflow. If the ICD Group button is not assigned, incoming calls will arrive at the INTERCOM or CO button.
  Even though an extension can have an ICD Group button for an incoming call distribution group that the extension has not been programmed to be included through system programming (→ Incoming Call Distribution Group Member [620]), the ICD Group button will not receive calls to that group.
- **Group FWD**
  The FWD feature can be assigned on an incoming call distribution group basis.
- **COS for Incoming Call Distribution Groups**
  Each incoming call distribution group is assigned a COS number. Group FWD to an outside party can be enabled or disabled for each COS. The COS for incoming call distribution groups is also used for the Internal Call Block feature; when an extension user calls an incoming call distribution group, the PBX checks the COS of the calling extension against the COS of the incoming distribution group (→ 1.1.2.2 Internal Call Block).

Feature Guide References

1.3.1.2 Call Forwarding (FWD)

1.18.2 Flexible Buttons

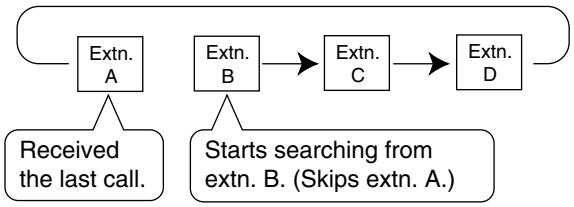
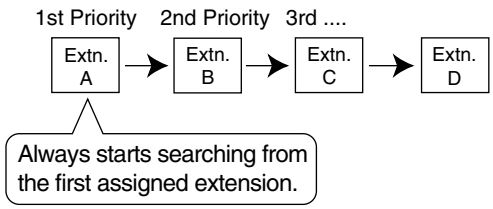
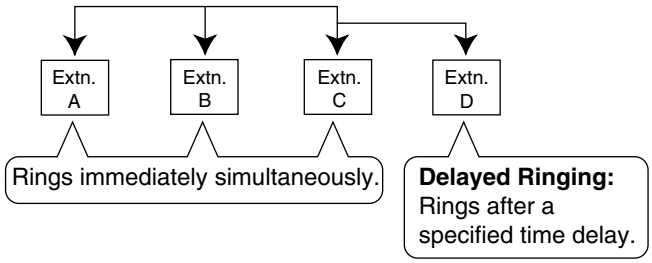
1.2.2.2 Group Call Distribution

Description

The Incoming calls directed to an incoming call distribution group are distributed to the member extensions using the distribution method until a preprogrammed number of extensions (agents) (→ Maximum Number of Agents [632]) are available to accept a call. When incoming calls exceed the number, the calls enter a queue (→ 1.2.2.3 Queuing Feature).

1. Distribution Method

There are three distribution methods which can be assigned to each incoming call distribution group (→ Incoming Call Distribution Group Distribution Method [624]).

Distribution Method	Description
Uniform Call Distribution (UCD)	Calls go to a different extension uniformly each time a call is received. Extensions are hunted in a circular way in the preprogrammed order for the group, starting at the extension after the extension which received the last call. 
Priority Hunting	An idle extension is searched for using the preprogrammed order for the group. 
Ring	All extensions in the group ring simultaneously. **Delayed Ringing:** Delayed ringing or no ringing can be programmed for each extension in the group (→ Incoming Call Distribution Group Delayed Ringing [621]). The call can be answered by pressing the flashing button even if no ring or a delayed time is set. 

2. Call Waiting for Incoming Call Distribution Group (Group Call Waiting)

When there are no available extensions in an incoming call distribution group, the group members can receive the Call Waiting tone. To use this feature:

- Select the Group Call Waiting mode through system programming. This determines the distribution method for waiting calls.
- Member extensions must assign the Call Waiting mode individually, or they will not be notified. (→ 1.1.3.3 Call Waiting)

[How the Group Call Waiting Feature Activates]

<KX-TDA100/KX-TDA200>

Programming Conditions		Result	
Group Call Waiting Mode	**Group Call Distribution Method**	**Group Call Waiting Distribution Method**	**Capable Telephone**
Distribution	UCD	UCD	PT/PS with idle ICD Group button
	Priority Hunting	Priority Hunting	
	Ring	Ring*	Any telephone
All	UCD/Priority Hunting/ Ring		

*: Delayed Ringing for PTs and SLTs is not available.

<KX-TDA30>

Programming Conditions		Result	
Group Call Waiting Mode	**Group Call Distribution Method**	**Group Call Waiting Distribution Method**	**Capable Telephone**
Distribution	UCD	UCD	PT/PS with idle ICD Group button
	Priority Hunting	Priority Hunting	
	Ring	Not available*	Any telephone
All	UCD/Priority Hunting/ Ring	Ring	

*: Incoming calls enter the queue immediately. Member extensions do not receive the Call Waiting tone.

[Example]

• Group Call Waiting mode: All • Group call distribution method for idle extensions: UCD	→ All extensions hear the Call Waiting tone (Ring).

[ICD Group Button for Group Call Waiting]

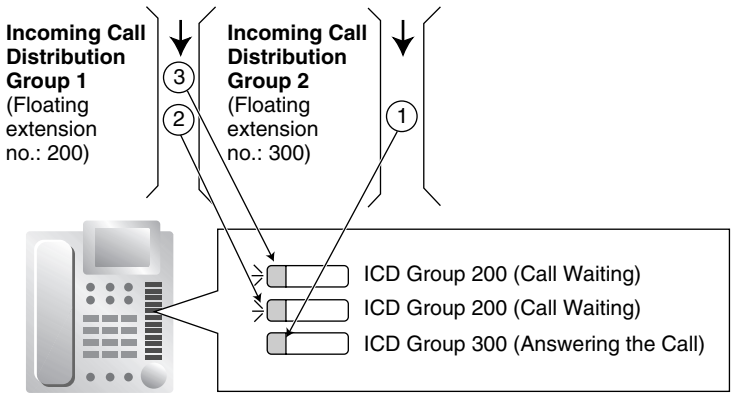
The way that the Group Call Waiting feature works depends on the Group Call Waiting Distribution method as follows:

a) Ring: The Group Call Waiting feature activates for all busy member extensions (even when the extensions do not have ICD Group buttons) simultaneously for only one incoming call — additional calls will wait in a queue.

b) UCD/Priority Hunting: The Group Call Waiting feature activates on an idle ICD Group button located on busy member extensions in a certain order. (This order depends on the type: UCD or Priority Hunting.) Calls will arrive at idle buttons until all ICD Group buttons are occupied — additional calls will wait in a queue.

Note

In the method b), if an extension has one or more ICD Group buttons for an incoming call distribution group and all the ICD Group buttons on the extension are occupied, the Group Call Waiting feature for the group will not work at the extension.


Incoming Call Distribution Group 1 (Floating extension no.: 200)
Incoming Call Distribution Group 2 (Floating extension no.: 300)
3
2
1
ICD Group 200 (Call Waiting)
ICD Group 200 (Call Waiting)
ICD Group 300 (Answering the Call)

3. **No Reply Redirection (UCD or Priority Hunting Method)**
   If a call received at a member extension is not answered within a preprogrammed time period (No Answer time), the call will be redirected to the next member extension. If there is no idle group member, the call queues at target extension until a group member becomes available.

Conditions

- **FWD/DND Extension**
  System programming for each incoming call distribution group is required to skip or ring the extension which has FWD or DND feature set. If it rings, the FWD/DND settings are ignored. (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND))
- Group Call Waiting feature cannot be used with the VIP Call feature (→ 1.2.2.4 VIP Call) and/or Wrap-up feature (→ 1.2.2.6 Log-in/Log-out). Because to use the VIP Call feature and/or Wrap-up feature, Call Waiting mode on each extension should be off.

1.2.2.3 Queuing Feature

Description

When a preprogrammed number of extensions (→ Maximum Number of Agents [632]) in an incoming call distribution group are busy, additional incoming calls can wait in a queue. The number of calls which can be in the waiting queue is programmable (→ Queuing Call Capacity [628]).
While calls are waiting in the queue, the calls are handled by the Queuing Time Table (→ Queuing Time Table [630]), which can be assigned for each time mode (day/lunch/break/night) (→ 2.2.4 Time Service). Each Queuing Time Table has a specified number of sequences. The following procedures are provided to make up a Queuing Time Table:

[Procedure Table]

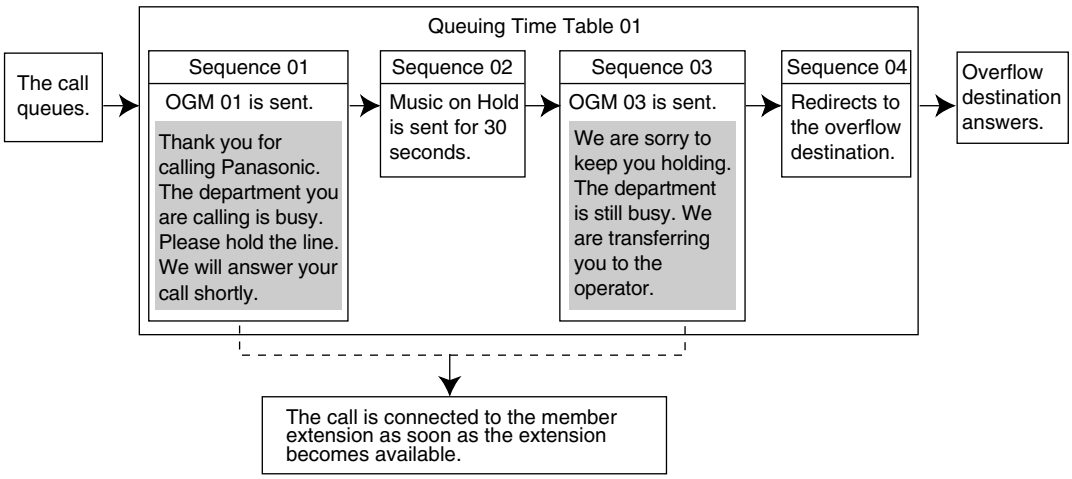
Command	Description	Condition
OGM a	An outgoing message (OGM) a (01–64 [with the KX-TDA100/KX-TDA200] or 01–32 [with the KX-TDA30]) is sent to the caller.	After the outgoing message (OGM), Music on Hold will be sent and redirects to the next sequence.
b × 5 s	Puts the caller in the waiting queue for b (01–16) × 5 seconds.	If an outgoing message (OGM) has not been sent to the caller, the caller hears a ringback tone. If an outgoing message (OGM) has been sent to the caller, the caller hears Music on Hold.
Sequence c	Redirects to sequence c (01–16).	None
Overflow	Redirects to overflow destination.	None
Disconnect	Disconnects the line.	None
None (No command)	Redirects to the next sequence.	If assigned on sequence 01, the Queuing Time Table will not be activated.

[Programming Example of Queuing Time Table]

Queuing Time Table No.	Sequence*1					
	Sequence 01	**Sequence 02**	**Sequence 03**	**Sequence 04**	**...**	**Sequence 16***2
01	OGM 01	6 × 5 s	OGM 03	Overflow		
02						
03						
:	:	:	:	:	:	:

*1: → Sequences in Queuing Time Table [631]
*2: The call will be disconnected if the destination is not determined after sequence 16.

Explanation for Queuing Time Table 01:


Queuing Time Table 01
The call queues.
Sequence 01
OGM 01 is sent.
Thank you for calling Panasonic. The department you are calling is busy. Please hold the line. We will answer your call shortly.
Sequence 02
Music on Hold is sent for 30 seconds.
Sequence 03
OGM 03 is sent.
We are sorry to keep you holding. The department is still busy. We are transferring you to the operator.
Sequence 04
Redirects to the overflow destination.
Overflow destination answers.
The call is connected to the member extension as soon as the extension becomes available.

Conditions

- **If the call is transferred to the incoming call distribution group and is handled by the Queuing Time Table:**
  Transfer Recall will not occur even if the Transfer Recall time expires.
- **Manual Queue Redirection**
  It is possible to redirect the longest waiting call in a queue to the overflow destination by pressing the Hurry-up button. (When the call is already ringing on any extension, the call is not redirected.)
  This feature is also known as Hurry-up Transfer.
- **Hurry-up Button**
  A flexible button can be customised as the Hurry-up button. The number of calls queuing before Manual Queue Redirection may be performed is programmable (→ Queuing Hurry-up Level [629]). The button shows the current status as follows:

Light Pattern	Calls in the Waiting Queue
Off	No queued call
Red on	At or under the assigned number for Hurry-up
Rapid red flashing	Over the assigned number for Hurry-up

Feature Guide References

1.2.2.5 Overflow Feature

1.18.2 Flexible Buttons

User Manual References

User Manual

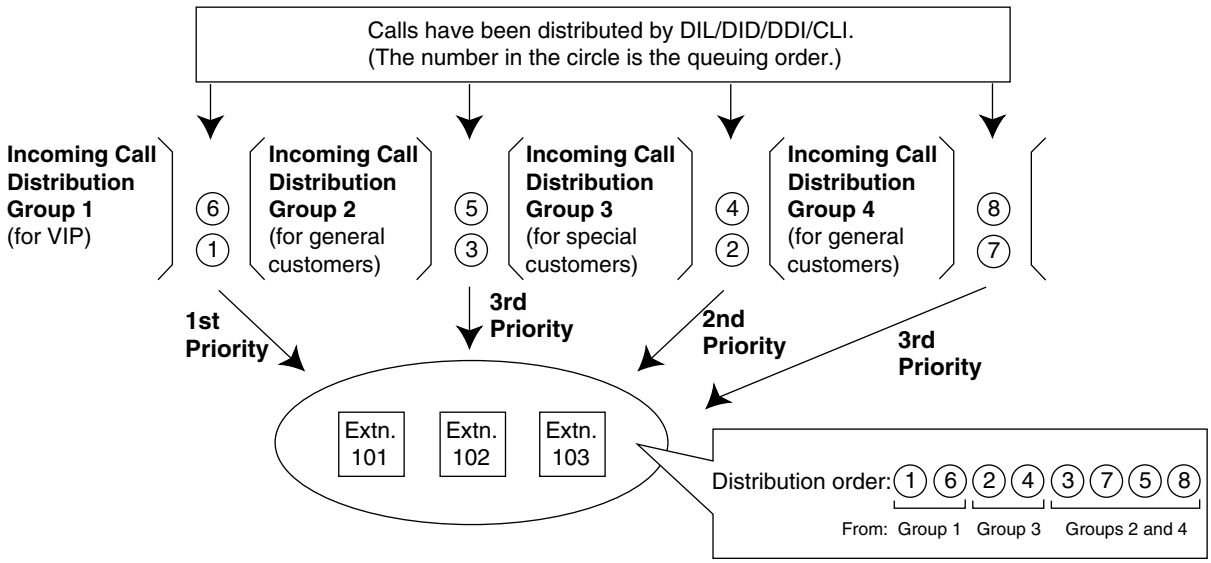
1.8.3 Forwarding a Waiting Call (Manual Queue Redirection)

1.2.2.4 VIP Call

Description

It is possible to assign a priority to incoming call distribution groups. If an extension belongs to multiple groups and the extension becomes idle, queuing calls in the groups will be distributed to the extension in priority order.
Each incoming call distribution group can enable or disable the VIP Call mode. When multiple groups enable the VIP Call mode, the incoming call distribution group with the lowest numbered group has the highest priority. When multiple groups disable the VIP Call mode, queuing calls are distributed to extensions uniformly.

[Example]
In the call centre, incoming call distribution groups 1 and 3 enable the VIP Call mode, while incoming call distribution groups 2 and 4 disable the VIP Call mode.



1.2.2.5 Overflow Feature

Description

When waiting calls exceed the waiting queue capacity (→ Queuing Call Capacity [628]), they may be redirected to a preprogrammed destination or busy tone may be sent to the callers by the following features:

1) Intercept Routing—Overflow in an Incoming Call Distribution Group

2) Busy on Busy

1. Intercept Routing—Overflow in an Incoming Call Distribution Group

The Intercept Routing—Overflow in an Incoming Call Distribution Group works in one of following conditions:

a) There is no space in the waiting queue.

b) The Queuing Time Table is not assigned and there are no extensions logged-in.

c) An Overflow command is assigned to the Queuing Time Table.

d) The Overflow time (→ Overflow Time [626]) expires.

e) Manual Queue Redirection is performed.

[Available Destination]

The overflow destinations can be assigned for each incoming call distribution group and each time mode (day/lunch/break/night) (→2.2.4 Time Service). The two different destinations can be assigned, one for the a) and b) (→ Destination When All Busy [627]), and one for c), d), and e) (→ Destination for Overflow Time Expiration [625]).

Destination	Availability
Wired Extension (PT/SLT/ISDN Extension/T1-OPX)	✔
PS	✔
Incoming Call Distribution Group	✔
PS Ring Group	✔
VM Group (DTMF/DPT)	✔
External Pager (TAFAS)	✔
DISA	✔
Analogue/ISDN Remote Maintenance	✔
Idle Line Access no. + Phone no.	✔
Trunk Group Access no. + Trunk Group no. + Phone no.	✔
Other PBX Extension (TIE with no PBX Code)	✔
Other PBX Extension (TIE with PBX Code)	✔

2. **Busy on Busy**

   The Busy on Busy feature works when the destination for the Intercept Routing—Overflow in an Incoming Call Distribution Group feature is not assigned in one of the following conditions:

   a) There is no space in the Waiting queue.

   b) The Queuing Time Table is not assigned and there are no extensions logged-in.

   [Example of a)]
   When the answering agent number (→ Maximum Number of Agents [632]) is "2", and the queuing call number (→ Queuing Call Capacity [628]) is "0":
   There are five assistants in the shop. If two of them are engaged on the phone, the next caller will hear busy tone to prevent the caller from thinking that there is no one in the shop or that the shop is closed.

Conditions

[Intercept Routing—Overflow in an Incoming Call Distribution Group]

- **If the Overflow time expires, and the overflow destination is unavailable:**

  a) If the trunk call arrives through the LCOT or T1 (LCOT/GCOT) card: (1) The line is disconnected when the call was once in a queue and an outgoing message (OGM) was sent to the call, or when the call reached an incoming call distribution group by using the DISA feature (→ 1.15.6 Direct Inward System Access (DISA)). (2) Except for the above, redirection is ignored and the Overflow timer activates again.

  b) If the call arrives through the other cards: Redirection is ignored and the Overflow timer activates again.

[Busy on Busy]

- If a trunk call arrives through the LCOT or T1 (LCOT/GCOT) card, busy tone will not be sent to the caller.

Feature Guide References

1.2.2.3 Queuing Feature

1.2.2.6 Log-in/Log-out

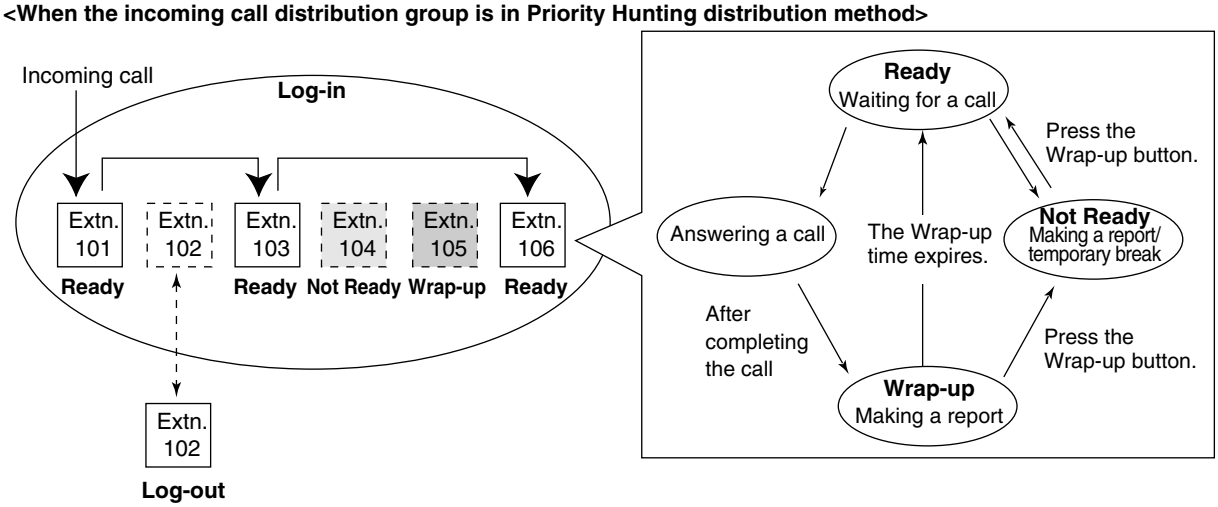
Description

Incoming call distribution group members can join (Log-in) or leave (Log-out) the groups manually.
They can leave the group temporarily when they are away from their desks, to prevent calls being sent to their extensions. They can return to the group when they are ready to answer calls.

Wrap-up:
While logged-in, a member extension can have a preprogrammed time period automatically for refusing calls after completing the last call (Wrap-up time). While Wrap-up timer is active, calls to all incoming call distribution groups to which the extension belongs will skip the extension so that he can make some report or etc.
Wrap-up mode can also be activated manually (Not Ready) by pressing the Wrap-up button.

[Log-in/Log-out and Wrap-up Status Example]

<When the incoming call distribution group is in Priority Hunting distribution method>



Conditions

- Generally, there should be at least one extension in a group that is logged-in. However, all extensions can log-out if permitted through system programming.
- **Log-in/Log-out Button**
  A flexible button can be customised as the Log-in/Log-out button with the following parameters:

Parameter	Usage	Light Pattern	
		Red on	Off
No parameter	Used with an ICD Group button, or with a floating extension number of an incoming call distribution group, or with ✳ (All).	—	—

Parameter	Usage	Light Pattern	
		Red on	Off
Floating extension number of a specified incoming call distribution group	Used to log-in or log-out from the specified incoming call distribution group.	Log-out Status	Log-in Status
✳ (All)	Used to log-in or log-out from all incoming call distribution groups to which you belong.	After Log-out Operation	After Log-in Operation

- If an ICD Group button is assigned, it also shows the log-in/log-out status of the corresponding group. The light pattern is the same as the Log-in/Log-out button which includes the group number.
- **Wrap-up Button**
  A flexible button can be customised as the Wrap-up button. It shows the current status as follows:

Light pattern	Status
Slow red flashing	Wrap-up
Red on	Not Ready
Off	Ready (Wrap-up mode cancel)

- When a PS in Wireless XDP Parallel Mode completes a call, neither the PS nor its wired telephone can have Wrap-up time. (→ 1.23.5 Wireless XDP Parallel Mode)
- **Automatic Log-out**
  A member extension may be logged-out automatically, if the Unanswered time expires a preprogrammed number of times consecutively. The number of consecutive unanswered calls can be assigned for each incoming call distribution group. If the extension is a member of more than one incoming call distribution group, the unanswered number is counted across all corresponding incoming call distribution groups. It is possible to return to the log-in mode manually.
  Automatic Log-out feature does not work for the extension in an incoming call distribution group in Ring distribution method (→ 1.2.2.2 Group Call Distribution).
- **Log-in/Log-out Monitor**
  The supervisor extension can monitor and control the log-in/log-out status of the incoming call distribution group members. (→ 1.2.2.7 Supervisory Feature)
- **Log-in/Log-out Information on SMDR**
  Log-in/Log-out information can be printed out on SMDR. (→ 1.24.1 Station Message Detail Recording (SMDR))

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.8.1 Leaving an Incoming Call Distribution Group (Log-in/Log-out, Wrap-up)

1.2.2.7 Supervisory Feature

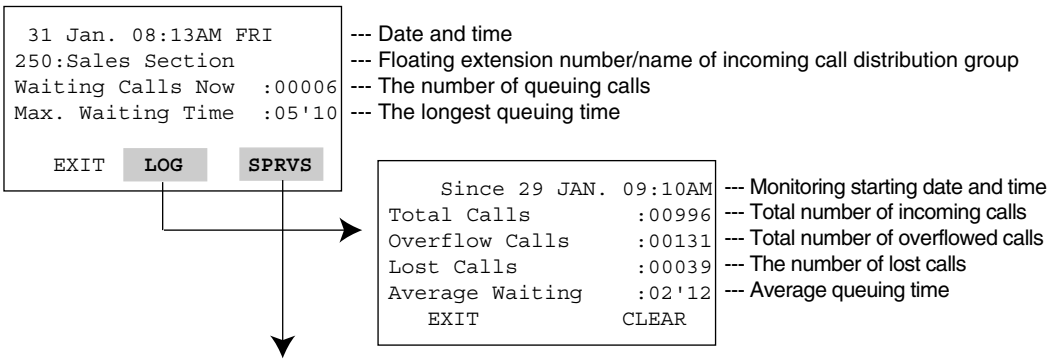
Description

An extension preprogrammed as a supervisor (supervisor extension) can monitor and control each member's status within the incoming call distribution group using a 6-line display PT.

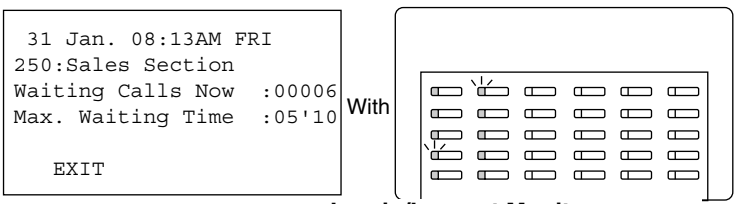
Feature	Description
Incoming Call Queue Monitor	The supervisor extension can monitor the status of an incoming call distribution group with the display.
Log-in/Log-out Monitor and Remote Control	**Monitor:** The supervisor extension can monitor the log-in/log-out status of the incoming call distribution group members through the corresponding DSS button light. **Remote Control:** The supervisor extension can change the status of the members by pressing the corresponding DSS button.

[Example]

<Incoming Call Queue Monitor Display>


31 Jan. 08:13AM FRI
250:Sales Section
Waiting Calls Now :00006
Max. Waiting Time :05'10
EXIT LOG SPRVS
--- Date and time
--- Floating extension number/name of incoming call distribution group
--- The number of queuing calls
--- The longest queuing time
Since 29 JAN. 09:10AM
Total Calls :00996
Overflow Calls :00131
Lost Calls :00039
Average Waiting :02'12
EXIT CLEAR
--- Monitoring starting date and time
--- Total number of incoming calls
--- Total number of overflowed calls
--- The number of lost calls
--- Average queuing time

<Log-in/Log-out Monitor/Remote Control Mode with DSS Button light>



Log-in/Log-out Monitor
DSS buttons of the incoming call distribution group members show their status.

Light pattern	Status
Green on	Log-in (Ready)
Slow Green Flashing	Log-in (Not Ready)
Red on	Log-out
Off	Extension in another incoming call distribution group

Log-in/Log-out Remote Control
Pressing the button changes the status as follows:

Status	Light pattern
Log-out	Red on
Log-in (Ready)	Green on

Conditions

- **Available Extension as a Supervisor Extension**
  - **a)** One supervisor extension can be assigned for each incoming call distribution group, but it need not belong to the group.
  - **b)** One extension can be the supervisor extension of more than one incoming call distribution group.
- **Available Paired DSS Console**
  The KX-T7640, KX-T7440, and KX-T7441 are available for this feature.
- **Accumulation Value Clear**
  Accumulation value (total incoming calls/total overflowed calls/lost calls/average queuing time) can be cleared manually. The date and time of clearing is saved and it is shown on the display (monitoring starting date and time). When the value exceeds 99999, before clearing, "*****" will be shown.
- **If a call to an incoming call distribution group is overflowed:**
  If the display is in idle status, it will change to monitor mode for the corresponding incoming call distribution group automatically.
  If the display is monitoring another incoming call distribution group, it will not change.
- **Other Features while in Monitor Mode**
  The supervisor extension can use other features (making calls, pressing the MESSAGE button, etc.) even while in monitor mode. When each operation is finished, his telephone returns to the queue monitor display.

User Manual References

User Manual

1.8.2 Monitoring and Controlling the Call Status of an Incoming Call Distribution Group (Incoming Call Distribution Group Monitor)

1.3 Call Forwarding (FWD)/Do Not Disturb (DND) Features

1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)

1.3.1.1 Call Forwarding (FWD)/Do Not Disturb (DND)—SUMMARY

Description

When an extension user cannot answer calls (e.g., busy, not at the desk), it is possible to forward or refuse the calls using the following features:

1) Call Forwarding (FWD)

2) Do Not Disturb (DND)

1. **FWD**
   Extensions and incoming call distribution groups can forward their incoming calls to preset destinations. (→ 1.3.1.2 Call Forwarding (FWD))

2. **DND**
   An extension user can send the tone to let the caller know he is not available. (→ 1.3.1.3 Do Not Disturb (DND))

Conditions

- **FWD and DND features apply to:**
  Intercom calls (including doorphone calls), and trunk calls (including a call from an extension that placed a trunk call on a consultation hold.)
- **FWD/DND Button**
  Both the FWD and DND features for the extension can be customised on a flexible button. Only one of the features will work at a time.
  Multiple types of FWD/DND buttons can be customised on an extension.
- **Group FWD Button**
  The FWD feature for the incoming call distribution group can be customised on a flexible button. Multiple types of Group FWD buttons can be customised on an extension.

  [FWD/DND Button and Group FWD Button Types]

Type		Description
FWD/DND for Extension	FWD/DND—Internal	Works for incoming intercom calls
	FWD/DND—External	Works for incoming trunk calls
	FWD/DND—Both	Works for all incoming calls

Type		Description
FWD for Incoming Call Distribution Group	Group FWD—Internal	Works for incoming intercom calls
	Group FWD—External	Works for incoming trunk calls
	Group FWD—Both	Works for all incoming calls

[Button Status—FWD/DND Button]
The FWD/DND button shows the current status as follows:

Light Pattern	Status (default)
Red on	FWD on
Slow red flashing	DND on
Off	FWD/DND off

The functions assigned to the "on" and "flashing" patterns can be changed through system programming.

[Button Status—Group FWD Button]
The Group FWD button shows the current status as follows:

Light Pattern	Status (default)
Red on	FWD on
Off	FWD off

[Mode Change]
When either the FWD or DND feature is assigned, pressing the FWD/DND button changes the on/off setting alternately. When both the features are assigned simultaneously, pressing the button changes the settings as follows:

Note

Pressing the FWD/DND button (fixed button) in idle status will produce one of the following results, selected through system programming:

When in FWD/DND Setting Mode:
Pressing the button will enter personal programming mode for the FWD/DND setting.

When in FWD/DND Cycle Switch Mode:
Pressing the button will cycle the settings as shown above in [Mode Change].

A FWD/DND button customised on a flexible button is always in FWD/DND Cycle Switch mode, and the mode cannot be changed.

- When intercom calls are set to be handled differently from trunk calls (forwarding type, forward destination, DND on/off), we recommend establishing buttons for both FWD/DND—Internal and FWD/DND—External, and/or Group FWD—Internal and Group FWD—External, because:

a) the light patterns of the FWD/DND—Both button (including FWD/DND button [fixed button]) and the Group FWD—Both button will indicate the setting for either trunk calls

or intercom calls, but not both.

Note

The FWD and DND icons on a PS display reflect the settings for trunk calls only.

b) pressing the FWD/DND—Both button (including FWD/DND button [fixed button]) or the Group FWD—Both button will not change the FWD or DND mode for intercom calls and trunk calls separately.

Feature Guide References

1.18.1 Fixed Buttons

1.18.2 Flexible Buttons

User Manual References

User Manual

3.1.2 Settings on the Programming Mode

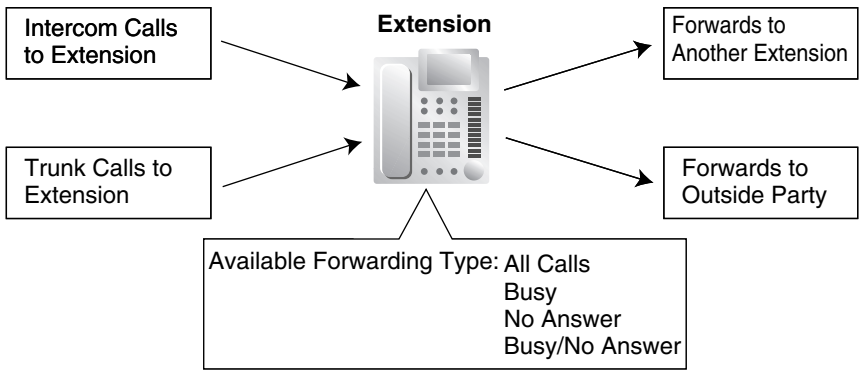
1.3.1.2 Call Forwarding (FWD)

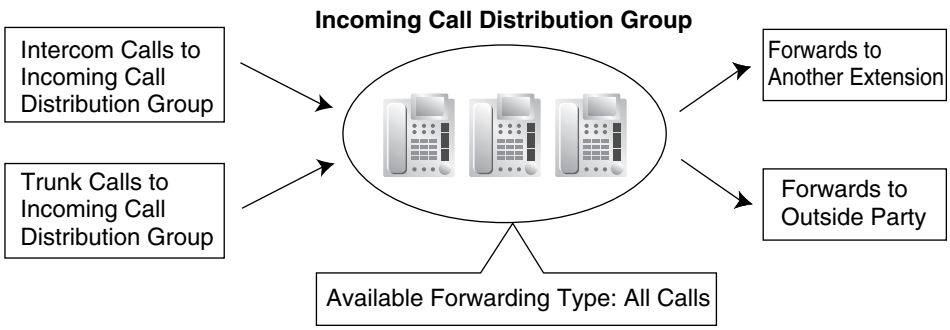
Description

Extensions and incoming call distribution groups can forward their calls to preset destinations. The circumstances under which the calls are forwarded are as follows:

Type	Circumstance
All Calls	Any time **Follow Me:** When an extension user fails to set this feature before leaving the desk, this feature can be set from the destination extension.
Busy	When the extension user's line is busy.
No Answer	When the extension user does not answer within a preprogrammed time (→ Call Forwarding—No Answer Time [605]).
Busy/No Answer	When the extension user's line is busy or the user does not answer within a preprogrammed time (→ Call Forwarding—No Answer Time [605]).

Depending on the type of incoming intercom or trunk calls, it is possible to set different destination for each.




Incoming Call Distribution Group
Intercom Calls to Incoming Call Distribution Group
Trunk Calls to Incoming Call Distribution Group
Forwards to Another Extension
Forwards to Outside Party
Available Forwarding Type: All Calls

[Available Destination]

Destination	Availability	Condition for Original Extension/Incoming Call Distribution Group
Wired Extension (PT/SLT/ISDN Extension/T1-OPX)	✔	Only available when FWD to extension is allowed through COS programming.*
PS	✔	
Incoming Call Distribution Group	✔	
PS Ring Group	✔	–
VM Group (DTMF/DPT)	✔	–
External Pager (TAFAS)	✔	–
DISA	✔	Only available for incoming trunk calls. Incoming intercom and doorphone calls cannot be forwarded to a DISA floating extension number.
Analogue/ISDN Remote Maintenance	✔	–
Idle Line Access no. + Phone no.	✔	Only available when FWD to trunk is allowed through COS programming.
Trunk Group Access no. + Trunk Group no. + Phone no.	✔	
Other PBX Extension (TIE with no PBX Code)	✔	–
Other PBX Extension (TIE with PBX Code)	✔	Only available when FWD to trunk is allowed through COS programming.

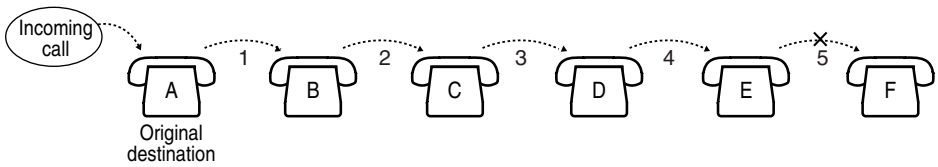
*: If an extension user cannot call certain extensions on a COS basis (→ 1.1.2.2 Internal Call Block), the FWD feature to the extension does NOT work.

Conditions

[General]

- **FWD for Trunk Calls/Intercom Calls**
  The FWD feature can be set for trunk calls, for intercom calls, or for both of them by the extension user.
- **FWD from Incoming Call Distribution Group (Group FWD)**
  COS programming determines the incoming call distribution groups that can use this feature.
- **FWD to Trunk**
  COS programming determines the extensions or incoming call distribution groups that can forward calls externally (→ Call Forwarding to Trunk [504]).
  The original extension's TRS/Barring and ARS still apply to the forwarded call.

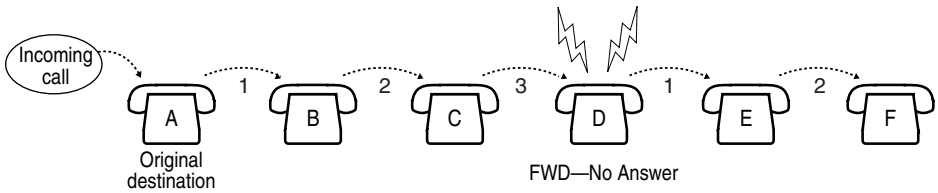
- **Trunk Call Duration**
  If a call between an extension user and an outside party, or between two outside parties is established, the call duration can be restricted by a system timer (→ Extension-to-Trunk Call Duration [472] and Trunk-to-Trunk Call Duration [473]). If the timer expires, the line will be disconnected. (→ 1.10.8 Trunk Call Limitation)
- **Multiple FWD**
  Calls can be forwarded up to four times. The following forwarding features are counted as Multiple FWD:
  - FWD—Busy or Busy/No Answer (in case a destination extension is busy), or All Calls
  - Idle Extension Hunting—Overflow
  - Intercept Routing—Busy/DND (in case a destination extension is busy or in DND mode)
  - Incoming Call Distribution Group—Overflow



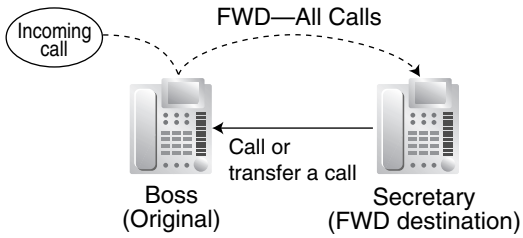
In the above illustration, forwarding stops at the extension E. However, forwarding can go farther in the following cases:

- If a destination extension rings, and then the call is redirected to the forward destination by the FWD—No Answer or Busy/No Answer feature.
- If a destination extension rings, and then the call is redirected to the intercept destination by the Intercept Routing—No Answer feature.
- If a call waits in a queue of an incoming call distribution group, and then the call is redirected to the overflow destination by the Queuing Time Table. (→ 1.2.2.3 Queuing Feature)

In the above cases, the forwarding counters reset to zero, and calls can be forwarded up to four times again from the applicable extension that occurred in the case above.



- **Boss & Secretary feature**
  It is possible to call the original extension from the destination extension regardless of the forward setting.



- **Message Waiting**
  While calls are forwarded, Message Waiting information is not forwarded. The Message button light turns on at the originally called extension. (→ 1.17.1 Message Waiting)
- **Idle Extension Hunting**
  Idle Extension Hunting applies to calls forwarded to a busy extension in an idle extension hunting group.

[All Calls and Busy]

- If the forward destination is not available to answer a call, this feature is cancelled and the original destination will ring for the following type of call:
  - Doorphone call
  - Trunk calls via the LCOT or T1 (LCOT/GCOT) cards

[No Answer and Busy/No Answer]

- **No Answer Time**
  The number of rings before the call is forwarded is programmable for each extension (→ Call Forwarding—No Answer Time [605]).

[Follow Me]

- This feature is only available when the original extension has disabled the "`Deny Remote Operation by Other Extension`" setting on a COS basis.

Feature Guide References

1.1.1.6 Intercept Routing
1.2.1 Idle Extension Hunting
1.2.2.5 Overflow Feature

User Manual References

User Manual

1.5.1 Forwarding Calls

1.3.1.3 Do Not Disturb (DND)

Description

An extension user can make use of the DND feature. If this feature is set, calls will not arrive at the extension, but arrive at other extension by using the Idle Extension Hunting feature (→ 1.2.1 Idle Extension Hunting) or the Intercept Routing—Busy/DND feature (→ 1.1.1.6 Intercept Routing). When a destination cannot be found, the calling extension will hear the DND tone, while the calling outside party will hear busy tone.

Conditions

- **DND for Trunk Calls/Intercom Calls**
  The DND feature can be set for trunk calls, for intercom calls, or for both of them by the extension user.
- **DSS button in DND Mode**
  The DSS button light will turn red if the assigned extension has set DND.
- **DND Override**
  An extension in DND mode can be called by other extension users who are allowed to override DND in their COS (→ DND Override [507]).
- **Paging DND**
  It is programmable whether the PBX pages extensions in DND mode through system programming. (→ 1.14.1 Paging)
- **Intercept Routing—Busy/DND**
  If a call arrives at the extension in DND mode, the call can be redirected to the preprogrammed destination by the Intercept Routing—Busy/DND feature.
- **Idle Extension Hunting**
  While searching for an idle extension within an idle extension hunting group, any extension which has DND set will be skipped. The call will go to the next extension in the group, not the Intercept Routing—Busy/DND destination.
- If (1) the trunk call via the LCOT or T1 (LCOT/GCOT) card arrives at the extension in DND mode and (2) the Intercept Routing—Busy/DND destination is not available and (3) there is no available extension in the idle extension hunting group, then the original extension in DND mode will ring.
- Calls from a doorphone arrive at the extension even when the extension is in DND mode.

User Manual References

User Manual

1.2.4 When the Dialled Line is Busy or There is No Answer

1.7.2 Refusing Incoming Calls (Do Not Disturb [DND])

1.4 Answering Features

1.4.1 Answering Features

1.4.1.1 Answering Features—SUMMARY

Description

An extension user can answer incoming calls by the following methods:

Destination	Feature	Description	Details in
At the own extension (PT only)	**Line Preference—Incoming**	A user can select the line seized when going off-hook.	• 1.4.1.2 Line Preference—Incoming
	Direct One-touch Answering	A user can answer an incoming call simply by pressing the flashing button.	–
	Hands-free Answerback	A user can receive a call automatically and establish a hands-free conversation.	• 1.4.1.4 Hands-free Answerback
At another extension	**Call Pickup—Directed/Group**	A user can pick up a specified extension's call or a call in a specified call pickup group.	• 1.4.1.3 Call Pickup

1.4.1.2 Line Preference—Incoming

Description

A PT user can select the method used to answer incoming calls from the following three line preferences:
Each of these line preferences can be assigned on each extension through personal programming (Preferred Line Assignment—Incoming).

Type	Description
No Line	Selects a line by pressing the desired Line Access button to answer an incoming call after you go off-hook.
Prime Line	Answers a call arriving at a Flexible CO or ICD Group button (on which the "Prime Line" is assigned) simply by going off-hook. This works even when multiple calls are received simultaneously.
Ringing Line (default)	Answers the longest ringing call at one's telephone simply by going off-hook when multiple calls arrive.

Conditions

[Prime Line]

- The priority of the incoming call is as follows:
  1) The call arriving at a button on which the "Prime Line" is assigned.
  2) The call arriving at the INTERCOM button.

User Manual References

User Manual

3.1.2 Settings on the Programming Mode

1.4.1.3 Call Pickup

Description

An extension user can answer a call ringing at any other extension.
The following types are available:

Type	Picking up Call Type
Directed	A specified extension's call.
Group	A call within a specified call pickup group.

Call Pickup Deny:
Preventing other extensions from picking up calls ringing at your extension is also possible.

Conditions

- **Call Pickup applies to:**
  Intercom, trunk, and doorphone calls
- **Internal Call Block**
  An extension which cannot call certain extensions on a COS basis (→ 1.1.2.2 Internal Call Block) also cannot pick up any calls ringing at those extensions.

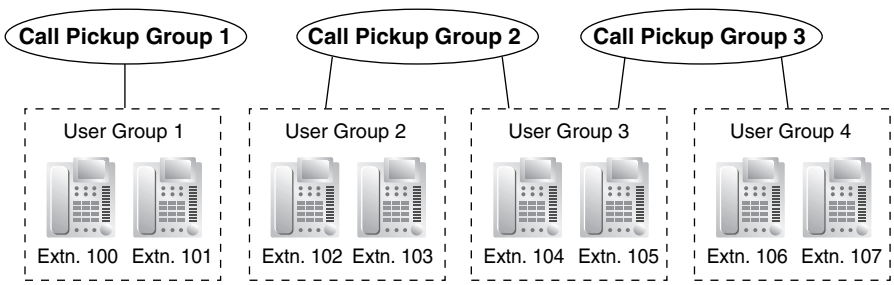
[Directed Call Pickup]

- A user can also pick up a specified extension's call (except an ISDN extension's call) by pressing the corresponding DSS button. COS programming determines the extensions that can use this feature.

[Group Call Pickup]

- A specified number of call pickup groups can be created (→ User Groups of a Pickup Group [650]), each of which consists of user groups. One user group can belong to several call pickup groups. (→ 2.2.2 Group)

  [Example]



User Manual References

User Manual

1.3.3 Answering a Call Ringing at Another Telephone (Call Pickup)

1.4.1.4 Hands-free Answerback

Description

A PT user with a speakerphone can talk to a caller without lifting the handset. If the user receives a call in Hands-free Answerback mode, a hands-free conversation is established in the following method:

Type	Answering Method
Intercom Call	Established immediately after a beep tone at the called extension and the caller hears a confirmation tone.
Trunk Call*	Established after a specified number of rings, a called extension hears a beep tone.

*: Including a call from an extension that placed a trunk call on a consultation hold.

Conditions

- **Hands-free Answerback applies to:**
  Intercom calls and trunk calls, including calls directed to an incoming call distribution group in UCD or Priority Hunting distribution method. (→ 1.2.2.2 Group Call Distribution)
- **Hands-free Answerback for Trunk Calls**
  System programming is required to use this feature.
- **Secret Monitor**
  The beep tone which the called party hears before answering can be eliminated through system programming.
- **Alternate Receiving/Calling Mode (Ring/Voice) Override**
  Hands-free Answerback overrides the Alternate Receiving mode preset on the telephone and the Alternate Calling mode from the caller.
- **Hands-free Answerback with Headset**
  The Hands-free Answerback feature can also be used with a headset.

Feature Guide References

1.5.3 Intercom Call

User Manual References

User Manual

1.3.2 Answering Hands-free (Hands-free Answerback)

1.5 Making Call Features

1.5.1 Predialling

Description

A display PT user can check and correct the number to be dialled on-hook. The call will be initiated after going off-hook.

Conditions

- **Storing the Predialled Number in the Personal Speed Dialling**
  The predialled number can be stored in the Personal Speed Dialling by pressing the AUTO DIAL/STORE button. (→ 1.6.1.5 Speed Dialling—Personal/System) In this case, the extension will enter into the personal programming mode automatically so that a name can be assigned for the stored number.

User Manual References

User Manual

3.1.2 Settings on the Programming Mode

1.5.2 Automatic Extension Release

Description

After going off-hook, if an extension user fails to dial any digits within a preprogrammed time period, the user will hear reorder tone. This operation applies to intercom calls only.

This feature is also known as Automatic Station Release.

Conditions

- A PT/PS user hears reorder tone for a preprogrammed time period, and then the PT/PS goes back to the idle status automatically. However, an SLT user will just hear reorder tone until he goes on-hook.
- **This feature works in one of the following cases:**
  When making an intercom call
  a) The first digit has not been dialled within a preprogrammed time period.
  b) After a digit is dialled, if subsequent digits are not dialled within a preprogrammed time period.

1.5.3 Intercom Call

Description

An extension user can call another extension user.

Conditions

- **Extension Number/Name Assignment**
  Extension numbers (→ Extension Number [003]) and names (→ Extension Name [004]) are assigned to all extensions. The assigned number and name are shown on display PTs during intercom calls.
- **DSS Button**
  It is possible to access another extension with one-touch by pressing the corresponding Direct Station Selection (DSS) button. A flexible button can be customised as a DSS button.
- **Call Directory—Extension Dialling**
  Display PT users can make a call by selecting stored names on the display.
- **Alternate Receiving—Ring/Voice**
  A PT user can select to receive intercom calls by ring tone or by voice, through personal programming (Alternate Receiving—Ring/Voice). If a user selects voice-calling, the calling party talks to the user immediately after a confirmation tone. Denying voice-calling can also be selected.
- **Alternate Calling—Ring/Voice**
  A caller can change the called party's preset call receiving method (ring tone or voice). By doing so, ring-calling is switched to voice-calling, or vice versa, at the called party. The called party may deny voice-calling.
- **Tone after Dialling**
  After dialling an extension number, a user will hear one of the following:

Type	Description
Ringback Tone	Indicates the called party is being called.
Confirmation Tone	Indicates the called party has set voice-calling.
Busy Tone	Indicates the called party is busy.
DND Tone	Indicates the called party has set DND.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.2.1 Basic Calling

1.2.6 Alternating the Calling Method (Alternate Calling—Ring/Voice)

1.11.2 Using the Directories

3.1.2 Settings on the Programming Mode

1.5.4 Trunk Call Features

1.5.4.1 Trunk Call Features—SUMMARY

Description

An extension user can use the following features when making a trunk call:

Feature	Description	Details in
Emergency Call	A user can dial the preprogrammed emergency numbers regardless of the restrictions imposed on the extension.	• 1.5.4.2 Emergency Call
Account Code Entry	A user can enter an account code to identify outgoing calls for accounting and billing purposes.	• 1.5.4.3 Account Code Entry
Pulse to Tone Conversion	A user can temporarily convert the Pulse mode to DTMF mode to access special services.	• 1.5.4.4 Dial Type Selection
Pause Insertion	A user can insert a preprogrammed Pause time into a dialling number by pressing the PAUSE button, or it is automatically inserted between the user-dialled code (e.g., Host PBX Access code or Special Carrier Access code) and the following digits.	• 1.5.4.6 Pause Insertion • 1.5.4.7 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX) • 1.5.4.8 Special Carrier Access Code

1.5.4.2 Emergency Call

Description

An extension user can dial the preprogrammed emergency numbers (→ Emergency Number [304]) after seizing a trunk regardless of the restrictions imposed on the extension.

Conditions

- A specified number of emergency numbers can be stored (some may have default values).
- Emergency numbers may be called even when:
  - in Account Code—Forced mode (→ 1.5.4.3 Account Code Entry)
  - in any TRS/Barring levels (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring))
  - after the preprogrammed call charge limit is reached (→ 1.8.2 Budget Management)
  - in Extension Lock (→ 1.8.3 Extension Lock)

1.5.4.3 Account Code Entry

Description

An account code is used to identify outgoing trunk calls for accounting and billing purposes. The account code is appended to the SMDR call record. Therefore, for example, the firm uses an account code for each client so that the firm can determine what calls were made for the client, and can submit a bill to the client according to the client's account code on the SMDR call record.
There are two methods of entering account codes as follows:
One of the methods is selected for each extension on a COS basis (→ Account Code Mode [508]).

Mode	Description
Option	A user can enter an account code if needed at any time desired.
Forced	A user must always enter an account code before seizing a trunk so as not to forget entering the code.

Conditions

- An account code can be stored into Memory Dialling (e.g., One-touch Dialling).
- **Account Button**
  A flexible button can be customised as the Account button. The Account button is used in place of the feature number for entering an account code. This button is useful because it can be used at any time, while the feature number entry is allowed only when hearing a dial tone before seizing a trunk.
- Account code entry after receiving a disconnection signal from a trunk must be done while hearing reorder tone. Otherwise the SMDR call record is output and entry becomes impossible afterwards.
- If the account code is entered more than once, the code entered last is printed out in the SMDR.
- Even in Forced mode, emergency numbers can be dialled out without an account code. (→ 1.5.4.2 Emergency Call)
- PT users can also enter an account code for incoming trunk calls during a conversation.
- **Verified Code Entry**
  To identify who made a trunk call for accounting and billing purpose, a verified code is used. This code can be used at any extension. (→ 1.8.6 Verified Code Entry)

Feature Guide References

1.18.2 Flexible Buttons

1.24.1 Station Message Detail Recording (SMDR)

User Manual References

User Manual

1.2.1 Basic Calling

1.5.4.4 Dial Type Selection

Description

The dialling mode (rotary or tone) can be selected for each analogue trunk through system programming (→ LCOT Dialling Mode [410]) regardless of originating extension (under contract with the telephone company).
There are the following modes:

Mode	Description
DTMF (Dual Tone Multi-Frequency)	The dialling signal from an extension is converted to tone dialling. DTMF signals are transmitted to the trunk.
Pulse Dial (Rotary)	The dialling signal from an extension is converted to rotary dialling. Rotary pulses are transmitted to the trunk.

Conditions

- **Pulse to Tone Conversion**
  It is possible for an extension user to temporarily convert the Pulse mode to DTMF mode so that the user can access special services such as computer-accessed long distance calling or voice mail services. To convert to the DTMF mode, wait for a preprogrammed time period (Default: 5 seconds) after the trunk is connected, or press ✳. This feature works only on trunks set to Pulse mode. DTMF mode cannot be changed to Pulse mode.
- It is possible to select a pulse rate for the trunk port which has been set to Pulse mode (→ LCOT Pulse Rate [411]). There are two pulse rates: Low (10 pps) and High (20 pps).
- It is possible to assign the minimum duration of the DTMF signal sent to the trunk port which has been set to DTMF mode (→ LCOT DTMF Minimum Duration [412]).

1.5.4.5 Reverse Circuit

Description

The circuit in the PBX can detect the reverse signal from the telephone company when an extension user is trying to make a trunk call. This detects the start (a called party goes off-hook) and end (the called party goes on-hook) of an outgoing trunk call. When an trunk call is received, the circuit can also detect the reverse signal after an outside caller goes on-hook. The duration of call can be verified on SMDR using this feature (→ 1.24.1 Station Message Detail Recording (SMDR)).
It is possible to select whether the PBX detects the reverse signal for outgoing trunk calls only, or for both outgoing and incoming trunk calls, or for no trunk call (disable the detection) through system programming (→ LCOT Reverse Circuit [415]).

1.5.4.6 Pause Insertion

Description

A preprogrammed Pause time (→ LCOT Pause Time [416]) will be inserted manually or automatically.
Manual Insertion: Pressing the PAUSE button.
Automatic Pause Insertion: A pause will be automatically inserted between the user-dialled codes below and the following digits.

a) Host PBX Access code (→ 1.5.4.7 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX))

b) Special Carrier Access code (→ 1.5.4.8 Special Carrier Access Code)

c) Second Dial Tone Waiting code

Conditions

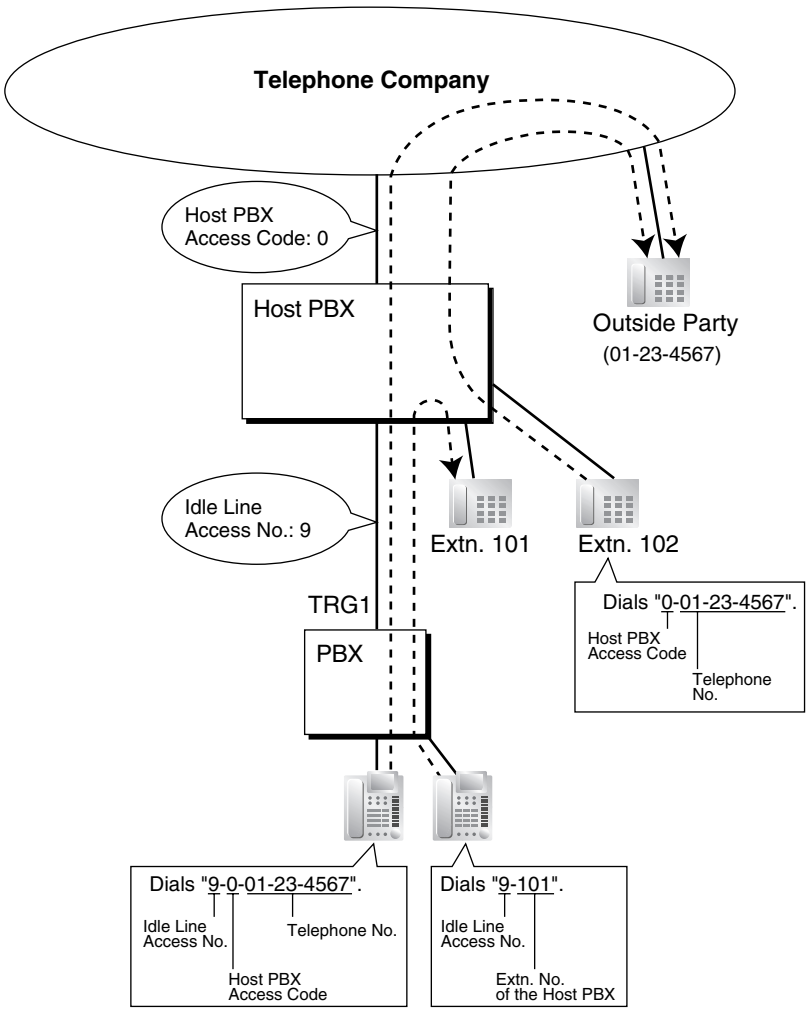
- The Pause time is programmable for each trunk.
- Pause can be stored in Memory Dialling.
- Pressing the PAUSE button while dialling a number inserts a pause for a preprogrammed time period.
- When a preprogrammed Second Dial Tone Waiting code is dialled after seizing a trunk, pauses are inserted a preprogrammed number of times after the code.

1.5.4.7 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)

Description

This PBX can be installed behind an existing host PBX. This is performed by connecting extension ports of the host PBX to trunk ports of this PBX. A Host PBX Access code assigned through system programming (→ Host PBX Access Code [471]) is required to access the telephone company from the host PBX. The Trunk Access number of the host PBX should be stored as a Host PBX Access code on a trunk group of this PBX basis.
A preprogrammed Pause time (→ LCOT Pause Time [416]) will be automatically inserted between the user-dialled Host PBX Access code and the following digits. (→ 1.5.4.6 Pause Insertion)

[Example]



Note: "0" should be assigned as a Host PBX Access code for trunk group (TRG) 1 of this PBX.

Conditions

- **TRS/Barring**
  TRS/Barring checks only the dialled telephone number excluding the Host PBX Access code when accessing the telephone company through the host PBX. (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring))
- **SMDR**
  The dialled number including the Host PBX Access code can be recorded on the SMDR when accessing the telephone company through the host PBX.
- To record on the SMDR only long distance calls (not local calls) originated via the specific trunk group, assign the long distance call code as a Host PBX Access code to the trunk group.

Feature Guide References

1.24.1 Station Message Detail Recording (SMDR)

1.5.4.8 Special Carrier Access Code

Description

If the PBX has access to multiple telephone companies, a Special Carrier Access code assigned through system programming (→ Special Carrier Access Code [303]) is required every time when a trunk call is made.
A preprogrammed Pause time (→ LCOT Pause Time [416]) will be automatically inserted between the user-dialled Special Carrier Access code and the following digits. (→ 1.5.4.6 Pause Insertion)

Conditions

- **TRS/Barring**
  TRS/Barring checks only the dialled telephone number excluding the Special Carrier Access code. (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring))
- **If this PBX is installed behind an existing host PBX:**
  A Special Carrier Access code and a Host PBX Access code should be assigned separately: these codes cannot be assigned together as one code. (→ 1.5.4.7 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX))

1.5.5 Seizing a Line Features

1.5.5.1 Seizing a Line Features—SUMMARY

Description

An extension user can select the line seized for making calls by the following methods:

Feature	Description	Details in
Line Preference—Outgoing	A user can select the line seized when going off-hook.	• 1.5.5.2 Line Preference—Outgoing
Trunk Access	A user can select the Trunk Access method every time he makes a trunk call.	• 1.5.5.3 Trunk Access

1.5.5.2 Line Preference—Outgoing

Description

A PT user can select the outgoing line he prefers to originate calls on, from the following line preferences, through personal programming (Preferred Line Assignment—Outgoing):

Feature	Description
Intercom	When an extension user goes off-hook, an extension line is selected automatically.
Idle Line	When an extension user goes off-hook, an idle trunk is selected automatically from the assigned trunk groups.
No Line	When an extension user goes off-hook, no line is selected. He must select the desired line to make a call.
Prime Line	When an extension user goes off-hook, the preset line is selected automatically. A prime line can be selected from the Line Access buttons: S-CO, G-CO, L-CO, ICD Group.

Conditions

- **Line Preference Override**
  A user can override the preset Line Preference temporarily by pressing the desired Line Access button or Memory Dialling button (e.g., One-touch Dialling) before going off-hook.
- To select Idle Line Preference, the trunk groups available to the extension should be programmed on a COS basis (→ Trunk Group Number [500]). Also trunk groups available for Idle Line Access should be assigned (→ Idle Line Access (Local Access) [103]).

User Manual References

User Manual

3.1.2 Settings on the Programming Mode

1.5.5.3 Trunk Access

Description

There are the following features to access a trunk.

Feature	Description	Accessing method
Idle Line Access (Local Access)	Selects an idle trunk automatically from the assigned trunk groups.	Dial the Idle Line Access number. Or press a L-CO button.
Trunk Group Access	Selects an idle trunk from the corresponding trunk group.	Dial the Trunk Group Access number and a trunk group number. Or press a G-CO button.
S-CO Line Access	Selects the desired trunk directly.	Dial the S-CO Line Access number and the trunk number. Or press the S-CO button.

Conditions

- COS programming determines the trunk groups available for making calls (→ Trunk Group Number [500]).
- Trunk numbers can be referred on a trunk port basis (→ LCOT/BRI Trunk Number Reference [409]).
- **Button Assignment**
  A flexible button can be customised as a G-CO, L-CO, or S-CO button as follows:

Type	Assignable parameter
Loop-CO (L-CO)	No parameter (All assigned trunk groups through system programming are applied.)
Group-CO (G-CO)	A trunk group is assigned.
Single-CO (S-CO)	A specified trunk is assigned.

  It is possible to assign trunks in the following ways:

  – The same trunk to the S-CO button and to a G-CO button
  – The same trunk group to more than one G-CO button
  – More than one L-CO button

  Dialling the Trunk Access number selects a CO button according to the priority: S-CO → G-CO → L-CO
- **Direct Trunk Access**
  By pressing an idle CO button, it automatically switches on the hands-free operation mode and allows a user to use On-hook Dialling. The user need not press the SP-PHONE button, MONITOR button, or lift the handset.
- **Group Hunting Order for Idle Line Access**
  An idle trunk is selected from the trunk groups assigned for Idle Line Access. If multiple trunk groups are available, the trunk group hunting sequence can be determined through system programming.

- **Trunk Hunting Order for Idle Line Access and Trunk Group Access**
  The trunk hunting sequence in a trunk group; from lowest numbered trunk, from highest numbered trunk or rotation, can be determined through system programming.
- A company name or customer name can be assigned on a trunk port basis (→ LCOT/BRI Trunk Name [401]) so that the operator or extension user can view the destination which the caller is trying to reach before answering.
- It is possible to identify the trunk port that has a trunk connected to it (→ LCOT/BRI Trunk Connection [400]). This prevents extension users from originating a call to a trunk which is not connected.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.2.1 Basic Calling

1.6 Memory Dialling Features

1.6.1 Memory Dialling Features

1.6.1.1 Memory Dialling Features—SUMMARY

Description

An extension user can store frequently dialled numbers in the PBX extension data and/or the PBX system data. A stored number is dialled automatically with a simple operation.

1. Features

Feature		Storing Method	Details in
One-touch Dialling		• Personal Programming • System Programming (PC Programming only)	• 1.6.1.2 One-touch Dialling
KX-T7710 One-touch Dialling (KX-TDA100/ KX-TDA200 only)		System Programming (PC Programming only)	• 1.6.1.3 KX-T7710 One-touch Dialling (KX-TDA100/KX-TDA200 only)
Last Number Redial (Outgoing Call Log)		The last dialled telephone number(s) is automatically stored.	• 1.6.1.4 Last Number Redial
Speed Dialling	**Personal**	• Personal Programming • Personal Operation with the Feature Number • System Programming (PC Programming only)	• 1.6.1.5 Speed Dialling—Personal/System
	System	System Programming	
Quick Dialling		System Programming (PC Programming only)	• 1.6.1.6 Quick Dialling
Hot Line		• Personal Programming • Personal Operation with the Feature Number • System Programming (PC Programming only)	• 1.6.1.7 Hot Line
Incoming Call Log		Incoming call information is automatically stored.	• 1.16.2 Incoming Call Log

2. Valid Input

Input	Display while Entering	Description
0–9/✶/#	0–9/✶/#	Stores the digits, ✶ and #.
PAUSE (Pause)	P	Stores a pause by pressing the PAUSE button. (→ 1.5.4.6 Pause Insertion)
FLASH/RECALL (Hooking)*	F	Stores a flash/recall signal (EFA mode) by pressing the FLASH/RECALL button at the beginning of the number. (→ 1.10.7 External Feature Access (EFA))
INTERCOM (Secret)*	[/]	Conceals all or part of the number by pressing the INTERCOM button at the beginning and at the end of the number to be concealed. It is programmable whether the concealed part will appear on the SMDR.
TRANSFER (Transfer)*	T	Stores a transfer command by pressing the TRANSFER button at the beginning of the number (used only for a One-touch Dialling). (→ 1.11.1 Call Transfer) **[Example]** Storing "T + 305"= Transferring a call to extension 305.

*: Available only when in the system/personal programming mode

[Example]
When storing the number "9-123-456-7890" and concealing the telephone number "123-456-7890",

Enter [9] → [INTERCOM] → [1][2][3][4][5][6][7][8][9][0] → [INTERCOM].

Notes

- It is possible to store a Memory Dialling feature number in the beginning of the Memory Dialling numbers.
- It is possible to store several feature numbers in one Memory Dialling location.

Conditions

- **Trunk Access by Memory Dialling**
  A specific Trunk Access number can be stored with the telephone number in Memory Dialling. However, if Memory Dialling is done after selecting a trunk, the stored Trunk Access number is ignored and the telephone number is sent using the selected trunk.

1.6.1.2 One-touch Dialling

Description

A PT user can access a person or feature by one-touch. This is activated by storing the number (e.g., extension number, telephone number, or feature number) in a One-touch Dialling button.

Conditions

- **One-touch Dialling Button**
  A flexible button can be customised as a One-touch Dialling button.
- **Full One-touch Dialling**
  There is no need to go off-hook before pressing the One-touch Dialling button.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.2.2 Easy Dialling

1.6.1.3 KX-T7710 One-touch Dialling (KX-TDA100/KX-TDA200 only)

Description

The Message button and One-touch buttons on all KX-T7710 telephones connected to the PBX can be customised at once through system programming. The same extension number, telephone number, or feature number will be assigned to the same buttons on each KX-T7710, useful for hotel room extensions or similar applications.

[Programming Example]

Button	Desired Number
Message	✳702 (Message Waiting [To Call Back])
One-touch Dial 01	100 (Front Desk)
One-touch Dial 02	✳7601 (Wake-up Call)
One-touch Dial 03	102 (Restaurant)
:	:

The Message button is programmed by default to call back a caller who left a message waiting indication, however, the Message button can be programmed to perform other features. The eight One-touch buttons have no default setting.

Conditions

- One-touch Dialling numbers for the KX-T7710 follow the flexible numbering plan. (→ 2.3.5 Flexible Numbering/Fixed Numbering)
- The KX-T7710 has two modes, NORMAL mode and PBX mode, selected by a switch on the telephone. This feature is available only when the KX-T7710 is in the PBX mode.
- This feature is available while hearing a dial tone.
- Please refer to the Quick Reference Guide of the KX-T7710 for additional information.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.2.2 Easy Dialling

1.6.1.4 Last Number Redial

Description

Every extension automatically saves the last external telephone number dialled to allow the same number to be dialled again.

Automatic Redial:
If Last Number Redial is performed in hands-free mode and the called party is busy, redialling will be automatically retried a preprogrammed number of times (→ Automatic Redial Repeat Times [205]) at preprogrammed intervals (→ Automatic Redial Interval [206]). The Redial Call No-answer Ring Duration time is programmable.
This feature is available only on certain PT models which have the SP-PHONE button.

Outgoing Call Log:
The last five dialled numbers are automatically stored at each extension. A display PT user can redial easily any of the stored numbers.

Conditions

- The memorised telephone number is replaced by a new one.
- If any dialling operation is done or an incoming call is answered during Automatic Redial, Automatic Redial is cancelled.
- Automatic Redial is not available for some countries/areas when using an analogue trunk.
- **Interrupt Redial**
  When the called party or the seized trunk is busy, it is possible to press the REDIAL button continuously until the called party or the trunk becomes idle. There is no need to go off-hook, before pressing the REDIAL button.
- **Storing the Call Log Information in Personal Speed Dialling**
  When storing the number and name from the call log into Personal Speed Dialling, the Idle Line Access number is automatically attached to the telephone number. (→ 1.6.1.5 Speed Dialling—Personal/System)
- **Outgoing Call Log Display by REDIAL Button (KX-TDA30 only)**
  Pressing the REDIAL button on a display DPT while on-hook can display the Outgoing Call Log. System programming is required for this operation.

User Manual References

User Manual

1.2.3 Redial

1.11.1 Using the Call Log

1.6.1.5 Speed Dialling—Personal/System

Description

An extension user can make calls using abbreviated dialling to frequently dialled numbers which are stored in the PBX extension data, or the PBX system data (→ System Speed Dialling Number [001]).

Personal Speed Dialling is also known as Station Speed Dialling.

Conditions

[General]

- Any number (e.g., telephone number, feature number) can be stored in a speed dialling number. A name can be assigned to each Personal Speed Dialling number through personal programming, and System Speed Dialling number (→ System Speed Dialling Name [002]).
- **Call Directory—Speed Dialling**
  Display PT users can make a call by selecting stored names on the display.

[Personal Speed Dialling]

- **Personal Speed Dialling Display Lock**
  An extension user can lock the Personal Speed Dialling number display to prevent other users from viewing the number. In this case, the Incoming and Outgoing Call Log information displays are also locked. An extension personal identification number (PIN) is required to use this feature. (→ 1.25.1 Extension Personal Identification Number (PIN))

[System Speed Dialling]

- **TRS/Barring Override by System Speed Dialling**
  It is possible to override the TRS/Barring using the System Speed Dialling (→ TRS/Barring Level for System Speed Dialling [509]). (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring))
- **System Speed Dialling Display by AUTO DIAL/STORE Button (KX-TDA30 only)**
  Pressing the AUTO DIAL/STORE button on a display DPT while on-hook can display the System Speed Dialling Directory.

User Manual References

User Manual

1.2.2 Easy Dialling

1.11.1 Using the Call Log

1.11.2 Using the Directories

3.1.2 Settings on the Programming Mode

3.3 Customising Your System (System Programming)

1.6.1.6 Quick Dialling

Description

An extension user can access a person or feature easily. This is enabled by storing the number (e.g., extension number, telephone number or feature number) for Quick Dialling.

Conditions

- Quick Dialling is convenient in the following cases:
  - Room service calls in a hotel
  - Calling another branch via the public network. The extension user needs to dial only the extension number of another branch.
- Quick Dialling numbers follow the flexible numbering plan.
  (→ 2.3.5 Flexible Numbering/Fixed Numbering)
- The storing example is as follows:

Location No.	Quick Dialling No.	Desired Number
Quick Dialling 01	110	9110 (Trunk Call)
Quick Dialling 02	5	3016 (Room Service)
Quick Dialling 03	2011	90123456789 (Another Branch)
:	:	:

User Manual References

User Manual

1.2.2 Easy Dialling

1.6.1.7 Hot Line

Description

An extension user can make an outgoing call by only going off-hook, if the user has previously stored the telephone number or the extension number.
If the Hot Line feature is set and the user goes off-hook, a dial tone is generated for a specified Waiting time assigned through system programming (→ Hot Line Waiting Time [204]) and then dialling starts. During the Waiting time the user can dial another party, overriding the Hot Line feature.

This feature is also known as Pickup Dialling.

Conditions

- **Capable Telephone**
  PT, SLT, T1-OPX, and PS

User Manual References

User Manual

1.2.2 Easy Dialling

1.7 Busy Line/Busy Party Features

1.7.1 Automatic Callback Busy (Camp-on)

Description

If the line is busy when a call is made, callback ringing will inform the caller when the line becomes free using this feature. After the extension answers the callback ringing, the dialled number is automatically redialled.

Conditions

- If the callback ringing is not answered within 10 seconds, the callback is cancelled.
- If the extension hears busy tone before dialling the telephone number, only the trunk or trunk group is reserved. After answering the callback ringing, the extension should dial the telephone number.
- An extension can set only one Automatic Callback Busy. The last setting is effective.
- Multiple extension users can set this feature to one trunk simultaneously.
  However, a maximum of four extension users can set this feature to one extension.
- The priority of the callback is the setting order.
- This feature cannot be used for calls to a VPS or an ISDN extension.

User Manual References

User Manual

1.2.4 When the Dialled Line is Busy or There is No Answer

1.7.2 Executive Busy Override

Description

Allows an extension user to interrupt an existing call to establish a three-party conference call.

Executive Busy Override Deny:
It is possible for extension users to prevent their calls from being intercepted by another extension user.

Conditions

- COS programming determines extension users who can use the Executive Busy Override (→ Executive Busy Override [505]) and set the Executive Busy Override Deny mode (→ Executive Busy Override Deny [506]).
- This feature does not work when the busy extension is in one of the following conditions:
  - **a)** Executive Busy Override Deny or Data Line Security (→ 1.10.5 Data Line Security) has been set.
  - **b)** While being monitored by another extension (→ 1.7.3 Call Monitor).
  - **c)** While receiving OHCA (→ 1.7.4.3 Off-hook Call Announcement (OHCA)) or Whisper OHCA (→ 1.7.4.4 Whisper OHCA).
  - **d)** During a Conference call (→ 1.13.1 Conference Features).
  - **e)** During a doorphone call (→ 1.15.1 Doorphone Call).
  - **f)** While Live Call Screening (LCS) or Two-way Record is activated (→ 1.22.3 Voice Mail DPT (Digital) Integration).
  - **g)** During Consultation Hold.

  (**Consultation Hold:** Allows an extension user to place a call on hold temporarily to perform Call Transfer, Conference, or Call Splitting.)
- This feature is not available for a trunk-to-trunk call via DISA.

User Manual References

User Manual

1.2.4 When the Dialled Line is Busy or There is No Answer

1.7.7 Preventing Other People from Joining Your Conversation (Executive Busy Override Deny)

1.7.3 Call Monitor

Description

Allows an extension user to listen to the busy extension user's existing conversation. The user can hear the conversation, but the user's voice is not heard. If desired, interrupting the call to establish a three-party conference call is available.

Conditions

- COS programming determines extension users who can use this feature.
- This feature is available only when the busy extension is in a conversation with another extension or outside party.
- This feature does not work when the busy extension is in one of the following conditions:
  - **a)** Executive Busy Override Deny (→ 1.7.2 Executive Busy Override) or Data Line Security (→ 1.10.5 Data Line Security) has been set.
  - **b)** While receiving OHCA (→ 1.7.4.3 Off-hook Call Announcement (OHCA)) or Whisper OHCA (→ 1.7.4.4 Whisper OHCA).
  - **c)** During a Conference call (→ 1.13.1 Conference Features).
  - **d)** During a doorphone call (→ 1.15.1 Doorphone Call).
  - **e)** While Live Call Screening (LCS) or Two-way Record is activated (→ 1.22.3 Voice Mail DPT (Digital) Integration).
  - **f)** During Consultation Hold.

  (**Consultation Hold:** Allows an extension user to place a call on hold temporarily to perform Call Transfer, Conference, or Call Splitting.)
- This feature stops when the busy extension user presses the following buttons during a conversation (→ 1.18.1 Fixed Buttons and 1.18.2 Flexible Buttons):
  - FLASH/RECALL button
  - HOLD button
  - TRANSFER button
  - CONF (Conference) button
  - DSS button
  - EFA button
  - Two-way Record button
  - Two-way Transfer button
  - One-touch Two-way Transfer button
  - Voice Mail (VM) Transfer button

User Manual References

User Manual

1.2.4 When the Dialled Line is Busy or There is No Answer

1.7.4 Second Call Notification to Busy Extension

1.7.4.1 Second Call Notification to Busy Extension—SUMMARY

Description

When attempting to call a busy extension (ringing or having a conversation), an extension user can send call waiting indication to the busy extension (**Call Waiting**). The notification receiving method depends on the called extension's personal setting and the telephone type:

Notification Receiving Method	Description	Details in
Call Waiting Tone	Sends the Call Waiting tone to the busy extension.	• 1.7.4.2 Call Waiting Tone
Off-hook Call Announcement (OHCA)	Talk with the busy extension using the built-in speaker and microphone of the called extension while the existing call is made using the handset.	• 1.7.4.3 Off-hook Call Announcement (OHCA)
Whisper OHCA	Give a message to the busy extension through the handset.	• 1.7.4.4 Whisper OHCA

Conditions

- Each extension user can choose to receive Call Waiting tone, OHCA, Whisper OHCA, or none of these.
- OHCA and Whisper OHCA are enabled or disabled by the COS of the calling extension.
- OHCA and Whisper OHCA do not work for some telephone types. In such cases, the Call Waiting tone will be sent to the called extension.

Calling Extension's OHCA COS Mode	Called Extension's Call Waiting Mode			
	OFF	ON		
	Cancel	Call Waiting Tone	OHCA	Whisper OHCA
Disable	Call Waiting disabled	Call Waiting tone	Call Waiting tone	Call Waiting tone
Enable	Call Waiting disabled	Call Waiting tone	OHCA (or Call Waiting tone)	Whisper OHCA (or Call Waiting tone)

- The notification receiving methods (Call Waiting tone, OHCA, and Whisper OHCA) are available when the called extension is having a conversation with other party. If not, the calling extension will be kept waiting until the called extension becomes available to receive the notification. Even while waiting, the calling extension will hear a ringback tone.
- If none of these notification receiving methods, Call Waiting tone, OHCA, or Whisper OHCA is set at the called party, the caller will hear reorder tone.

Feature Guide References

1.1.3.3 Call Waiting

User Manual References

User Manual

1.2.4 When the Dialled Line is Busy or There is No Answer

1.7.4.2 Call Waiting Tone

Description

When an extension user attempts to call a busy extension (ringing or having a conversation), the Call Waiting tone can be sent to the called extension to let him know another call is waiting.

Conditions

- This feature only works if the called extension has activated Call Waiting. If it is activated, the calling extension will hear a ringback tone.
- Call Waiting tone can be selected (Tone 1 or Tone 2) through personal programming (Call Waiting Tone Type Selection).

User Manual References

User Manual

3.1.2 Settings on the Programming Mode

1.7.4.3 Off-hook Call Announcement (OHCA)

Description

An extension user can talk with a busy extension through the built-in speaker and microphone of the called party's PT. If the existing call is using a handset, a second conversation is made using the speakerphone and microphone so that the called extension can talk to both parties.

Conditions

- COS programming determines which extensions can use this feature.
- This feature is available when the called extension uses one of the following telephones:
  - KX-T7625, KX-T7630, KX-T7633, KX-T7636
  - KX-T7536
  - KX-T7436
  - KX-T7235 (except KX-T7235G/FR/SL/NE)
- If the KX-T7235G/FR/SL/NE are connected to the PBX, the OHCA feature for the KX-T7235 should be disabled through system programming.
- The OHCA feature cannot be used in the following cases:
  - **(a)** COS or called extension's telephone type is not available for this feature.
  - **(b)** The called extension (DPT) is connected to a PC (PC Console or PC Phone) via the USB Module.
  - **(c)** The called extension (DPT) is in the Digital XDP connection. (KX-TDA30 only)

  The Call Waiting tone is sent to the called extension. (→ 1.7.4.2 Call Waiting Tone)
- While an extension is receiving OHCA, if the extension user places the current trunk call on hold or transfers the current intercom call or trunk call, OHCA will become disabled and the calling extension will start to hear a ringback tone.
- While an extension is receiving OHCA, if the extension user places the current intercom call on hold, the called extension can talk to the calling extension through the handset.

1.7.4.4 Whisper OHCA

Description

An extension user can give a message to a busy extension through the handset.

Conditions

- COS programming determines which extensions can use this feature.
- This feature is available when the calling and called extension use one of the following telephones:
  - KX-T7600 series
  - KX-T7500 series
  - KX-T7400 series (except KX-T7451)
- If the Whisper OHCA feature cannot be used due to COS or telephone type, the Call Waiting tone will be sent to the called extension. (→ 1.7.4.2 Call Waiting Tone)
- If the called extension does not use a KX-T7600, KX-T7500, or KX-T7400 series telephone but forces Whisper OHCA, the announcement may be heard by the other party.
- It is possible to enable Whisper OHCA on any telephone. However, it may not work properly. (e.g., The voice may be heard by the other party.)
- While an extension is receiving Whisper OHCA, if the extension user places the current trunk call on hold or transfers the current intercom call or trunk call, the Whisper OHCA will become disabled and the calling extension will start to hear a ringback tone.
- While an extension is receiving the Whisper OHCA, if the extension user places the current intercom call on hold, the called extension can talk to the calling extension through the handset.

1.8 Toll Restriction (TRS)/Call Barring (Barring) Features

1.8.1 Toll Restriction (TRS)/Call Barring (Barring)

Description

TRS/Barring can prohibit an extension user from making certain trunk calls by COS programming. It is applied when the user goes off-hook, a trunk is seized and then a dialled number is sent to the trunk.
Each COS is programmed to have a TRS/Barring level for each time mode (day/lunch/break/night).
There are seven levels available. Level 1 is the highest level and the level 7 is the lowest. That is, level 1 allows all trunk calls and level 7 prohibits all trunk calls. Levels 2 through 6 are used to restrict calls by combining preprogrammed Denied and Exception Code Tables.

Denied Code Tables

An outgoing trunk call made by an extension with a level between 2 and 6 is first checked against the applicable Denied Code Tables. If the leading digits of the dialled number (not including the Trunk Access number) are not found in the table, the call is made. There are five Denied Code Tables for Levels 2 through 6 respectively.
Complete every table by storing numbers that are to be prohibited. These numbers are defined as denied codes.

Exception Code Tables

These tables are used to override a programmed denied code. A call denied by the applicable Denied Code Tables is checked against the applicable Exception Code Tables, and if a match is found, the call is made.
There are five Exception Code Tables for Levels 2 through 6 respectively.
Complete every table by storing numbers that are exceptions to the denied codes. These numbers are defined as exception codes.

TRS/Barring Override by System Speed Dialling

If the call is made using System Speed Dialling, the call can override the TRS/Barring. Each COS is programmed to have a TRS/Barring level for System Speed Dialling (→ TRS/Barring Override by System Speed Dialling [300]).
Once this feature is set, it permits all extension users to make System Speed Dialling calls with the level for System Speed Dialling. Any extension which sets Extension Lock can also make a call using System Speed Dialling.

TRS/Barring Level

TRS/Barring level is determined by the applicable levels of Denied Code Tables and Exception Code Tables.

	Denied Code Tables*1	Exception Code Tables*2
Level 1	Not Programmable	Not Programmable
Level 2	Table for Level 2	Tables for Levels 2 through 6
Level 3	Tables for Levels 2 and 3	Tables for Levels 3 through 6
Level 4	Tables for Levels 2 through 4	Tables for Levels 4 through 6
Level 5	Tables for Levels 2 through 5	Tables for Levels 5 through 6
Level 6	Tables for Levels 2 through 6	Tables for Level 6
Level 7	Not Programmable	Not Programmable

*1: → TRS/Barring Denied Code [301]
*2: → TRS/Barring Exception Code [302]

Explanation:

Level 1: Allows all calls.

Level 2: Denies codes stored in the Denied Code Table for Level 2 except the codes stored in Exception Code Tables for Levels 2 through 6.

Level 3: Denies codes stored in the Denied Code Tables for Levels 2 and 3 except the codes stored in Exception Code Tables for Levels 3 through 6.

-
-
-

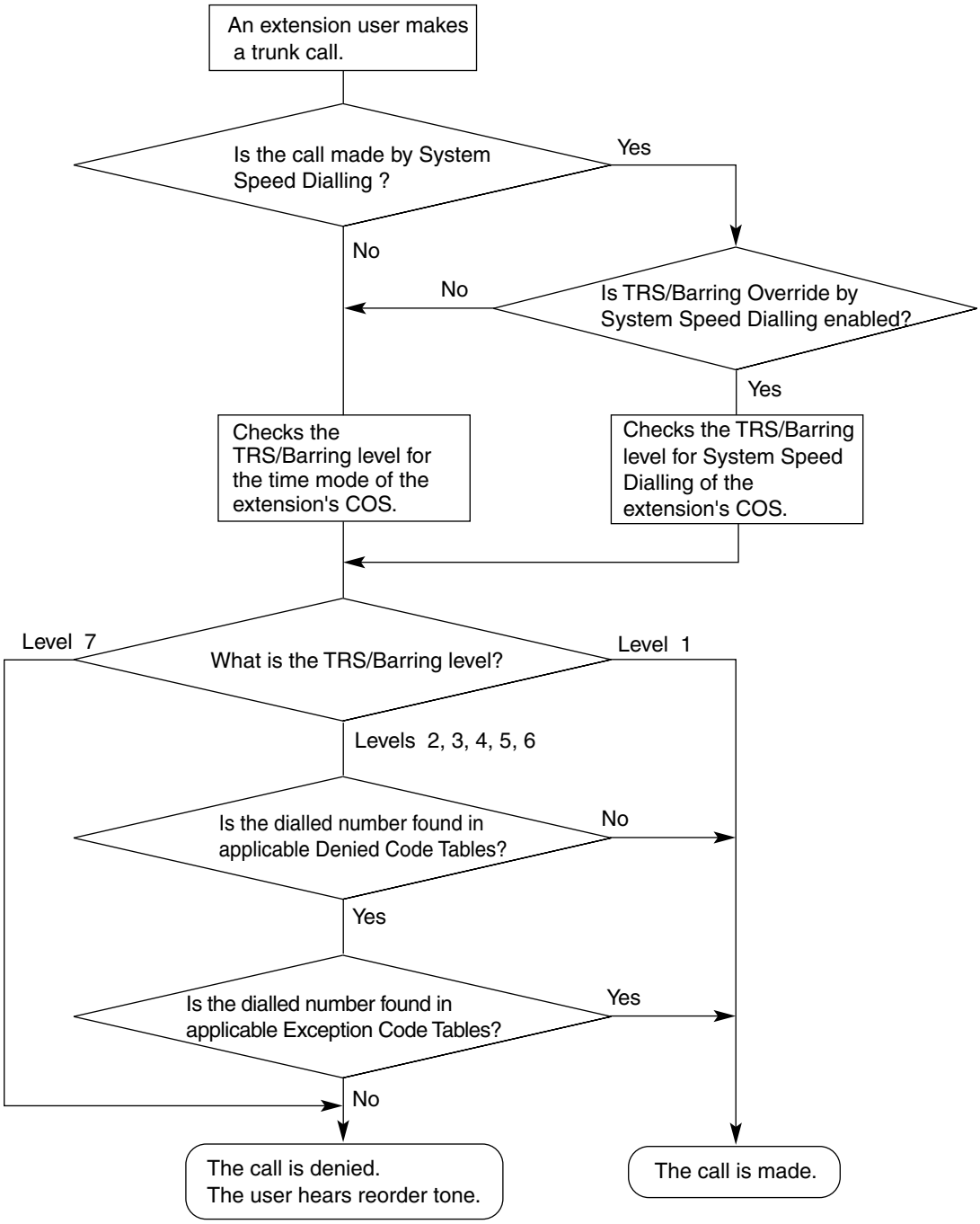
Level 7: Allows intercom calls only.

[Programming Example]

COS No.	Level for Time Mode*1				Level for System Speed Dialling*2
	Day	Lunch	Break	Night	
1	1	1	1	6	1
2	2	2	2	6	1
:	:	:	:	:	:

*1: → TRS/Barring Level [501]
*2: → TRS/Barring Level for System Speed Dialling [509]

[Flowchart]


An extension user makes a trunk call.
Is the call made by System Speed Dialling ?
Yes
No
No
Is TRS/Barring Override by System Speed Dialling enabled?
Yes
Checks the TRS/Barring level for the time mode of the extension's COS.
Checks the TRS/Barring level for System Speed Dialling of the extension's COS.
Level 7
What is the TRS/Barring level?
Level 1
Levels 2, 3, 4, 5, 6
Is the dialled number found in applicable Denied Code Tables?
No
Yes
Is the dialled number found in applicable Exception Code Tables?
Yes
No
The call is denied.
The user hears reorder tone.
The call is made.

Conditions

WARNING
The software contained in the TRS/Barring feature to allow user access to the network must be upgraded to recognise newly established network area codes and exchange codes as they are placed into service.
Failure to upgrade the premises PBXs or peripheral equipment to recognise the new codes as they are established will restrict the customer and the customer's employees from gaining access to the network and to these codes.
KEEP THE SOFTWARE UP-TO-DATE WITH THE LATEST DATA.

- A COS should be assigned for each extension (→ Class of Service [602])
- TRS/Barring checks are applied to the following:
  - ARS
  - Trunk Access (Idle Line/Trunk Group/S-CO Line)
- It is programmable whether "✳" or "#" is checked by the TRS/Barring. This is useful in preventing unauthorised calls which could be possible through certain telephone company exchanges.
- It is programmable whether TRS/Barring checks the digits dialled after the External Feature Access during a trunk call. (→ 1.10.7 External Feature Access (EFA))
- **Host PBX Access Code/Special Carrier Access Code**
  TRS/Barring checks for numbers dialled with a Host PBX Access code (→ 1.5.4.7 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)) or a Special Carrier Access code (→ 1.5.4.8 Special Carrier Access Code) in the following cases:

Type	Stored		Not stored
	Found	Not found	
Host PBX Access Code	Deletes the code. A TRS/Barring check is carried out on the following digits.	The call is made (excepted from TRS/Barring).	TRS/Barring checks the whole number.
Special Carrier Access Code	Deletes the code. A TRS/Barring check is carried out on the following digits.	TRS/Barring checks the whole number.	TRS/Barring checks the whole number.

- **ARS**
  If ARS is applied to a dialled number, TRS/Barring will check the user-dialled number (not the modified number by ARS). In this case, a Host PBX Access code and/or a Special Carrier Access code will not be checked.
- **Dialling Digit Restriction during Conversation**
  The dialling of digits can be restricted while engaged on a received trunk call. If the number of dialled digits exceeds the preprogrammed limitation, the line will be disconnected.
- A TRS/Barring level can be changed by some features. The priority of features, when multiple features are used, is as follows:
  1) Dial Tone Transfer (→ 1.8.4 Dial Tone Transfer)
  2) Budget Management (→ 1.8.2 Budget Management)
  3) TRS/Barring Override by System Speed Dialling

4) Walking COS/Verified Code Entry
(→ 1.8.5 Walking COS, 1.8.6 Verified Code Entry)

5) Extension Lock

Feature Guide References

1.5.5.3 Trunk Access

1.6.1.5 Speed Dialling—Personal/System

1.8.3 Extension Lock

1.9.1 Automatic Route Selection (ARS)

2.2.4 Time Service

1.8.2 Budget Management

Description

Limits the telephone usage to a preprogrammed budget on an extension basis. If the amount of the call charge reaches the limit, an extension user cannot make further trunk calls. An extension assigned as the manager may increase the limit or clear the amount of the call charge.

Conditions

- If the limit is reached, TRS/Barring Level 7 is applied. (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring))
- **Budget Management for Verified Call**
  If an extension user make a trunk call with a verified code, the call charge will be counted on the verified code (not the extension). (→ 1.8.6 Verified Code Entry) The limit of the call charge can be assigned on each verified code.
- The pay tone service or ISDN Advice of Charge (AOC) service is required for this feature.
- It is possible to select whether to disconnect the line after an alarm tone or only to send an alarm tone when the amount of the call charge reaches the preprogrammed limit during a conversation.

User Manual References

User Manual

3.2.2 Manager Programming

1.8.3 Extension Lock

Description

An extension user can change the TRS/Barring level of the telephone (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring)) so that other users cannot make inappropriate trunk calls. An extension personal identification number (PIN) is used to unlock the telephone (→ 1.25.1 Extension Personal Identification Number (PIN)).

This feature is also known as Electronic Station Lockout.

Conditions

- This feature also restricts changing the FWD destination. (→ 1.3.1.2 Call Forwarding (FWD))
- **Remote Extension Lock**
  Overrides Extension Lock. If an extension assigned as the manager sets Remote Extension Lock on an extension that has already been locked by the extension user, the user cannot unlock it. If a manager extension unlocks an extension that has been locked by the extension user, the extension will be unlocked.
  This feature is also known as Remote Station Lock Control.
- **TRS/Barring Level**
  COS programming determines the TRS/Barring level for Extension Lock (→ TRS/Barring Level for Extension Lock [510]).

User Manual References

User Manual

1.5.3 Preventing Other People from Using Your Telephone (Extension Lock)

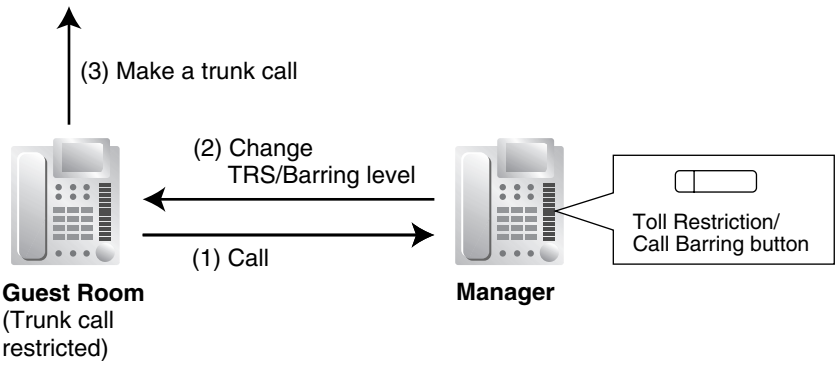
2.1.1 Extension Control

1.8.4 Dial Tone Transfer

Description

An extension assigned as the manager can change the TRS/Barring level (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring)) for the extension user temporarily. After that, the extension user can make his call.

[Example] An extension user can call a manager to release the restriction on outgoing call (e.g., international calls).



Conditions

- The changed level effects just one time.
- **Toll Restriction/Call Barring Button**
  A manager extension must store the desired TRS/Barring level in the Toll Restriction/Call Barring button. A flexible button can be customised as the Toll Restriction/Call Barring button.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

2.1.3 Restriction Level Control (Dial Tone Transfer)

1.8.5 Walking COS

Description

An extension user can use another extension telephone with the COS assigned on his own extension temporarily. The extension accesses his own extension from another telephone and operates features with the same conditions which are assigned his own extension. It is one of the remote operations. An extension personal identification number (PIN) is required to use this feature (→ 1.25.1 Extension Personal Identification Number (PIN)).

Conditions

- The following items are applied:
  - COS programming
  - Budget Management (→ 1.8.2 Budget Management)
  - Itemised Billing code for ARS (→ 1.9.1 Automatic Route Selection (ARS))
  - Extension number recorded on the SMDR (→ 1.24.1 Station Message Detail Recording (SMDR)).
- **Walking COS through DISA**
  This feature is also available through the DISA. (→ 1.15.6 Direct Inward System Access (DISA))

Feature Guide References

2.2.1 Class of Service (COS)

User Manual References

User Manual

1.2.7 Calling without Restrictions

1.8.6 Verified Code Entry

Description

An extension user can make a trunk call with a verified code from his own or any other extension to change the TRS/Barring level (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring)) or to identify the call for accounting and billing purposes. A verified code personal identification number (PIN) is required to use this feature.

Conditions

WARNING

When a third party discovers the personal identification number (PIN) (verified code PIN/ extension PIN) of the PBX, there will be a risk that they will make fraudulent phone calls using the telephone line, and the cost will be charged to your own account. In order to avoid this problem, we strongly recommend the following points:

a) Carefully maintain the secrecy of the PIN.

b) Specify a complicated PIN as long and random as you can make it.

c) Change the PIN frequently.

- The following items are applied:
  - COS programming
  - Budget Management
  - Itemised Billing code for ARS
  - The ✳ + verified code recorded on the SMDR instead of the extension number (→ 1.24.1 Station Message Detail Recording (SMDR)).

- **Verified Code Entry through DISA**
  This feature is also available through the DISA. (→ 1.15.6 Direct Inward System Access (DISA))
- **Verified Code PIN**
  A verified code PIN must be assigned for each verified code through system programming (→ Verified Code Personal Identification Number (PIN) [122]) or through manager programming.
- **Verified Code PIN Lock**
  If the wrong PIN is entered incorrectly three times, the line will be disconnected. If the incorrect PIN is entered a preprogrammed number of times successively, the PIN against the verified code will be locked. Only an extension assigned as the manager can unlock it. In this case, the PIN will be unlocked and cleared.

- **Budget Management for Verified Calls**
  A limit of the call charge can be assigned on each verified code.

  [Example of Verified Codes and Their Programming]

Location No.	Verified Code*1	User Name*2	Verified Code PIN	COS No.*3	Itemised Billing Code for ARS	Budget Management
0001	1111	Tom Smith	1234	1	2323	5000Yen
0002	2222	John White	9876543210	3	4545	3000Yen
:	:	:	:	:	:	:

*1: → Verified Code [120]
*2: → Verified Code Name [121]
*3: → Verified Code COS Number [123]

Feature Guide References

1.8.2 Budget Management
1.9.1 Automatic Route Selection (ARS)
2.2.1 Class of Service (COS)
2.2.6 Manager Features

User Manual References

User Manual

1.2.7 Calling without Restrictions
3.2.2 Manager Programming

1.9 Automatic Route Selection (ARS) Features

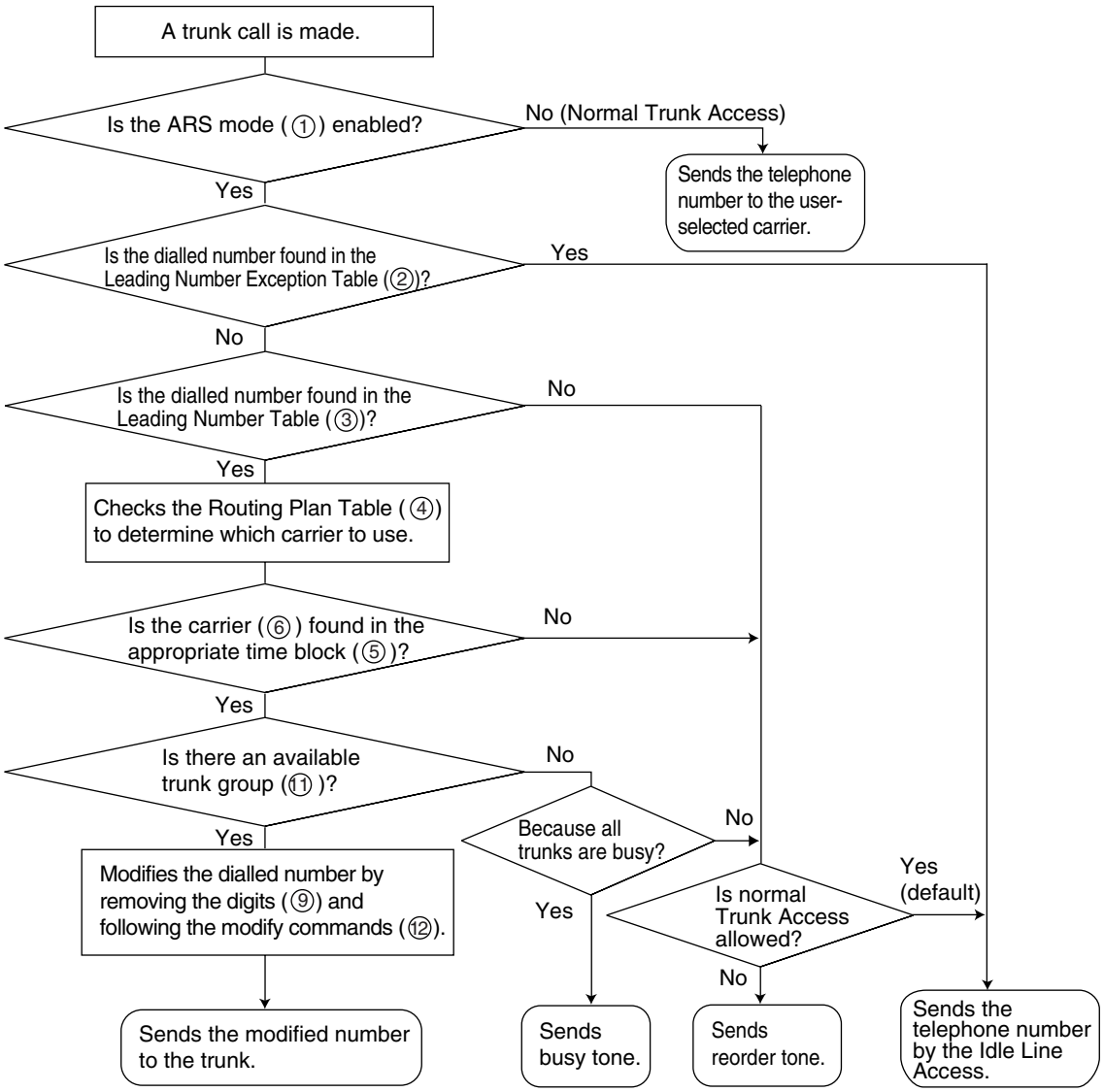
1.9.1 Automatic Route Selection (ARS)

Description

ARS automatically selects the carrier available at the time an outgoing trunk call is made according to preprogrammed settings. The dialled number will be checked and modified to connect the appropriate carrier.

[Carrier Selection Procedure Flowchart]

The numbers ⓧ in the flowchart correspond to the [Programming Procedures] on the following pages.


A trunk call is made.
Is the ARS mode (①) enabled?
No (Normal Trunk Access)
Sends the telephone number to the user-selected carrier.
Yes
Is the dialled number found in the Leading Number Exception Table (②)?
Yes
No
Is the dialled number found in the Leading Number Table (③)?
No
Yes
Checks the Routing Plan Table (④) to determine which carrier to use.
Is the carrier (⑥) found in the appropriate time block (⑤)?
No
Yes
Is there an available trunk group (⑪)?
No
Yes
Because all trunks are busy?
No
Yes
Modifies the dialled number by removing the digits (⑨) and following the modify commands (⑫).
Is normal Trunk Access allowed?
Yes (default)
No
Sends the modified number to the trunk.
Sends busy tone.
Sends reorder tone.
Sends the telephone number by the Idle Line Access.

[Programming Procedures]

1. **ARS Mode ① Assignment** (→ ARS Mode [320])
   It is possible to select whether ARS operates when an extension user makes a call using any Idle Line Access method or when an extension user makes a call using any Trunk Access method. (→ 1.5.5.3 Trunk Access)

2. **Leading Number Exception Table ② Assignment**
   Store the telephone numbers that will avoid using the ARS feature.

   ② ARS Leading Number Exception Table

Location No.	Leading No. Exception*1
001	033555
002	06456
:	:

   *1: → ARS Exception Number [325]

3. **Leading Number Table ③ Assignment**
   Store the area codes and/or telephone numbers as leading number that will be routed by the ARS feature. In this table, the Routing Plan (refer to "4. Routing Plan Table ④ Assignment") is selected for each number.
   The additional (remain) number of digits must be assigned only when "#", for example, is needed after a dialled number. The "#" is added after the assigned number of digits of dialled number (excluding a leading number).

   ③ ARS Leading Number Table

Location No.	Leading No.*1	Additional (Remain) No. of Digits	Routing Plan Table No.*2
0001	039	7	1
0002	03	0	4
0003	0444	5	5
:	:	:	:

   *1: → ARS Leading Number [321]
   *2: → ARS Routing Plan Table Number [322]

   If a dialled number matches a leading number, the number will be modified according to the corresponding Routing Plan Table and the modified number will be sent to the trunk when the assigned additional (remain) number of digits are dialled or the Inter-digit time expires.
   If a dialled number matches multiple leading number entries, the leading number entry with the lowest numbered location will have priority.

[Example]

Dialled Number	Corresponding Routing Plan Table No.	Description
039-123-4567	1	"039" is found in location 0001 and seven digits (assigned additional [remain] number of digits in location 0001) were dialled. The Routing Plan Table 1 is selected just after the seventh digit.
039-654-321	1	"039" is found in location 0001 and the Inter-digit time expired before the seventh digit is received. The Routing Plan Table 1 is selected after the Inter-digit time expired.
038	4	"03" is found in two locations (locations 0001 and 0002), so the PBX waits for the next digit "8". "038" is not found in any location, then "03" (location 0002) is selected. The Routing Plan Table 4 is selected.
03	4	The Inter-digit time expired after receiving "3". "03" is found in location 0002. The Routing Plan Table 4 is selected after the Inter-digit time expired.

4. Routing Plan Table ④ Assignment

Arrange the time schedule as desired and store the carrier priority.

Time Table ⑤

As the best carrier may vary with the day of the week and the time of day, four time blocks (Time-A through D) can be programmed for each day of the week.

Carrier Priority ⑥

Assign the appropriate carrier (refer to "5. Carrier Table ⑦ Assignment") and their priority in each time block. The carrier is selected in the entry order (the order in which entries are listed).

③ ARS Leading Number Table

Location No.	Leading No.	Additional (Remain) No. of Digits	Routing Plan Table No.
0001	03	8	1
:	:	:	:

④ ARS Routing Plan Table

Routing Plan Table 1					
⑤ Time Table*1			⑥ Carrier*2		
			Priority 1	Priority 2	...
SUN	Time-A	9:00	1 (A telecom)	4 (D telecom)	···
	Time-B	12:00	1 (A telecom)	2 (B telecom)	···
	Time-C	15:00	1 (A telecom)	2 (B telecom)	···
	Time-D	21:00	3 (C telecom)	1 (A telecom)	···
:	:	:	:	:	···
SAT	Time-A	9:00	3 (C telecom)	2 (B telecom)	···
	Time-B	12:00	3 (C telecom)	1 (A telecom)	···
	Time-C	15:00	3 (C telecom)	1 (A telecom)	···
	Time-D	21:00	3 (C telecom)	2 (B telecom)	···

*1: → ARS Routing Plan Time Table [330]
*2: → ARS Routing Plan Table (1–16) [331–346]

5. Carrier Table ⑦ Assignment

A specified number of carriers can be programmed. Assign the following items for each Carrier Table:

Carrier Name ⑧: Assign the carrier name.

Removed Number of Digits ⑨: Assign the number of digits to remove from the beginning of the user-dialled number.

Carrier Access Code ⑩: Assign the code to access the carrier.

Trunk Group ⑪**:** Assign the trunk groups which connect to each carrier.

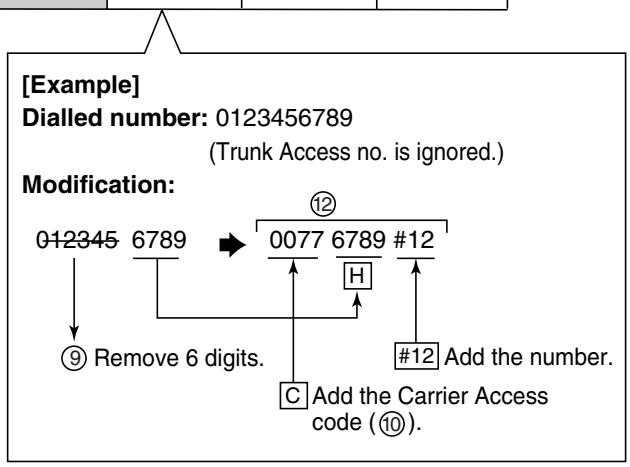
Modify Command ⑫**:** Assign the commands to modify the dialled number to access the carrier.

[Command Explanation]

Command	Description
Number	Add the number.
C	Add the Carrier Access code.
P	**Analogue Line:** Insert a pause. **ISDN/T1/E1 Line:** Insert a pause and change to tone (DTMF) signal.
A	Add the Authorisation code for a tenant (⑬).
G	Add the Authorisation code for a trunk group (⑭).
I	Add the Itemised Billing code (⑮).
H	Add the dialled number after the digits are removed (Home position).

[Programming Example]

⑦ Carrier Table	1	2	••
⑧ Carrier Name*1	A telecom	B telecom	••
⑨ Removed Number of Digits*2	6	0	••
⑩ Carrier Access Code*3	0077	0088	••
⑪ Trunk Group*4	1, 2, 3	1, 2	••
⑫ Modify Command	CH#12	CH	••



*1: → ARS Carrier Name [350]
*2: → ARS Removed Number of Digits for Carrier Access [352]
*3: → ARS Carrier Access Code [353]
*4: → ARS Trunk Group for Carrier Access [351]

6. Optional Assignment

Authorisation Code for a Tenant ⑬
An Authorisation code can be assigned for each carrier and each tenant.

Authorisation Code for a Trunk Group ⑭
An Authorisation code can be assigned for each trunk group and each carrier.

Itemised Billing Code ⑮
An Itemised Billing code can be assigned for each extension and for each verified code. If a call is not made from an extension (e.g., DISA or TIE) and no verified code is used, the Itemised Billing code assigned in the location 1 of the verified code will be used.

Conditions

WARNING
The software contained in the ARS feature to allow user access to the network must be upgraded to recognise newly established network area codes and exchange codes as they are placed into service.
Failure to upgrade the premises PBXs or peripheral equipment to recognise the new codes as they are established will restrict the customer and the customer's employees from gaining access to the network and to these codes.
KEEP THE SOFTWARE UP-TO-DATE WITH THE LATEST DATA.

- **The Dialled Number on the SMDR**
  It is possible to choose to print either the user-dialled number or the modified number on the SMDR through system programming. (→ 1.24.1 Station Message Detail Recording (SMDR))
- **ARS Data Download/Upload**
  It is possible to download or upload the following ARS data to the PBX using PC programming:
  - ③ ARS Leading Number Table
  - ④ ARS Routing Plan Table

  This is useful when a carrier has changed the call charge, and the updated data can be used for multiple customers.
- A TRS/Barring check is done before ARS is applied. (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring))

1.10 Conversation Features

1.10.1 Hands-free Operation

Description

A PT user can talk to another party without lifting the handset. Pressing specific buttons (e.g., REDIAL) automatically activate hands-free mode.

Conditions

- **PTs with the MONITOR Button**
  PTs with the MONITOR button can only dial in hands-free mode and cannot be used for hands-free conversations.

1.10.2 Off-hook Monitor

Description

A PT user can let others listen to the user's conversation through the built-in speaker, during a conversation using the handset.

Conditions

- **Capable Telephone**
  KX-T7400 series and KX-T7500 series (display PT only), and all KX-T7600 series telephones
  To enable this feature, system programming is required. If disabled, hands-free conversation is performed instead.

User Manual References

User Manual

1.4.7 Letting Other People Listen to the Conversation (Off-hook Monitor)

1.10.3 Mute

Description

During a conversation, a PT user can disable the speaker microphone or the handset microphone to consult privately with others while listening to the other party on the phone through the built-in speaker or the handset receiver. The user can hear the other party's voice during Mute. Only your voice will be muted.

Conditions

- This feature is available with all PTs that have the AUTO ANS/MUTE button.

User Manual References

User Manual

1.4.6 Mute

1.10.4 Headset Operation

Description

This PBX provides headset compatible PTs. A PT user can talk to another party without lifting the handset.
For connection and operation, refer to the Operating Instructions for the headset.

This feature is also known as Handset/Headset Selection.

Conditions

- A headset is an option.
- If the headset mode is on, pressing the SP-PHONE button activates the headset, not the built-in speaker.
- To set the headset mode on a DPT, use personal programming (Headset Operation) or press the Headset button. To set headset mode on an APT, use the handset/headset selector provided on the set and/or on the headset.
- **Headset Button**
  A flexible button on a DPT can be customised as the Headset button. The Headset button can be also assigned to a flexible button on an APT, but the button does not work.
- **Answer/Release Button**
  A flexible button can be customised as an Answer button or a Release button. Such buttons are useful for headset operation. It is possible to answer an incoming call by pressing an Answer button. While hearing the Call Waiting tone during a conversation, pressing an Answer button enables one to answer the second call by placing the current call on hold. Pressing a Release button enables one to disconnect the line during or after conversation, or to complete a Call Transfer.
- It is possible to switch the headset mode to the hands-free mode or vice versa during a conversation by pressing the Headset button.
- Headset users cannot use the following features:
  - Automatic Redial (→ 1.6.1.4 Last Number Redial)
  - Receiving OHCA (→ 1.7.4.3 Off-hook Call Announcement (OHCA))
  - Receiving Whisper OHCA (→ 1.7.4.4 Whisper OHCA)

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.3.5 Using the ANSWER/RELEASE Button
1.4.8 Using the Headset (Headset Operation)
3.1.2 Settings on the Programming Mode

1.10.5 Data Line Security

Description

Once Data Line Security is set on an extension, communication between the extension and the other party is protected from signals such as Call Waiting, Hold Recall and Executive Busy Override. Extension which has data devices connected (e.g., fax machine) may set this feature to maintain secure data transmission against tones or interruptions from other extensions during communication.

User Manual References

User Manual

1.7.9 Protecting Your Line against Notification Tones (Data Line Security)

1.10.6 Flash/Recall/Terminate

Description

The FLASH/RECALL button (Flash/Recall mode or Terminate mode) or Terminate button (Terminate mode) is used when a PT user disconnects the current call and originates another call without hanging up first. It performs as going on-hook and then going off-hook.
[Explanation of Each Mode]
Flash/Recall Mode: Disconnects the line and the extension user hears the dial tone from the line used last. For example, if a trunk call is disconnected, the extension user will hear a new dial tone from the telephone company.
Terminate Mode: Disconnects the line and the extension user hears the dial tone determined by the Line Preference—Outgoing setting. (→ 1.5.5.2 Line Preference—Outgoing)

Conditions

- **FLASH/RECALL Button Mode**
  One of the following modes can be selected for each extension through system programming:
  - Flash/Recall mode
  - Terminate mode
  - External Feature Access (EFA) mode. (→ 1.10.7 External Feature Access (EFA))
- **Terminate Button**
  A flexible button can be customised as the Terminate button.
- **Disconnect Time (Only for Flash/Recall Mode)**
  The amount of time between successive accesses to the same trunk is programmable for each trunk port (→ LCOT Disconnect Time [418]).
- This feature outputs an SMDR call record (→ 1.24.1 Station Message Detail Recording (SMDR)), restarts the call timer, inserts the automatic pause, and checks the TRS/Barring level (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring)) again.
- If the ARS mode is on (→ 1.9.1 Automatic Route Selection (ARS)) or if a trunk call is established through the INTERCOM button or ICD Group button, the Terminate mode will be always effective even though the Flash/Recall mode has been set.

Feature Guide References

1.18.2 Flexible Buttons

1.10.7 External Feature Access (EFA)

Description

An extension user can have access to the features of a host PBX or the telephone company, such as Transfer, Hold, Call Waiting, etc. A flash/recall signal is sent to the host PBX or the telephone company. This is only available on trunk calls (including Host PBX Access [→ 1.5.4.7 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)]).

Conditions

- **Flash/Recall Time**
  The Flash/Recall time can be assigned for each trunk port (→ LCOT Flash/Recall Time [417]).
- **EFA Button**
  A flexible button can be customised as the EFA button.
- Pressing the FLASH/RECALL button which is set to EFA mode (→ 1.10.6 Flash/Recall/Terminate) or the EFA button performs this feature. It is also possible to perform this feature by entering the feature number when the current call is placed on a Consultation Hold.
  (**Consultation Hold:** Allows an extension user to place a call on hold temporarily to perform Call Transfer, Conference, or Call Splitting.)

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.9.2 If a Host PBX is Connected

1.10.8 Trunk Call Limitation

Description

Trunk calls are limited by the following features:

Feature	Description
Extension-to-Trunk Call Duration	If a call between an extension user and an outside party is established, the call duration can be restricted by a system timer selected for each trunk group (→ Extension-to-Trunk Call Duration [472]). Both parties will hear an alarm tone 15 seconds before the time-limit at five second intervals. If the timer expires, the line will be disconnected. COS programming determines to enable or disable this feature (→ Trunk Call Duration Limitation [502]). Whether this feature is applied to outgoing calls only or all calls is also programmable.
Trunk-to-Trunk Call (except Unattended Conference Call) Duration	If a call between two outside parties is established, the call duration can be restricted by a system timer selected for each trunk group (→ Trunk-to-Trunk Call Duration [473]). Both parties will hear an alarm tone 15 seconds before the time-limit at five second intervals. If the timer expires, the line will be disconnected. The timer is applied for a trunk call which is made via the PBX. If both call is made by the extension (e.g., a trunk call made by an extension is transferred to an outside party), the shorter limit will apply to the call.
Budget Management	When the preprogrammed call charge limit is reached, an extension user will start to hear an alarm tone three times at five second intervals and the user cannot make further trunk calls. Whether the line is disconnected after the alarm tone is programmable. (→ 1.8.2 Budget Management)
Dialling Digit Restriction during Conversation	While engaged on a received trunk call, the dialling of digits can be restricted. If the number of dialled digits exceeds the limitation, the line will be disconnected.

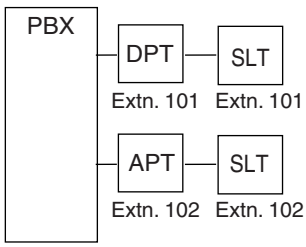
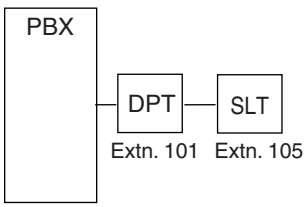
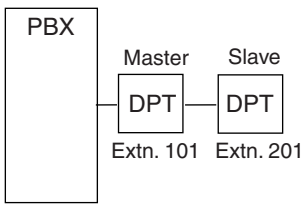
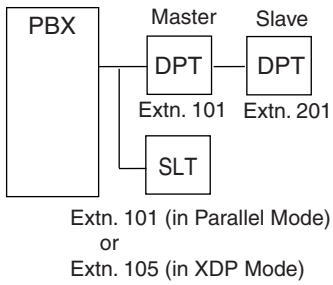
Conditions

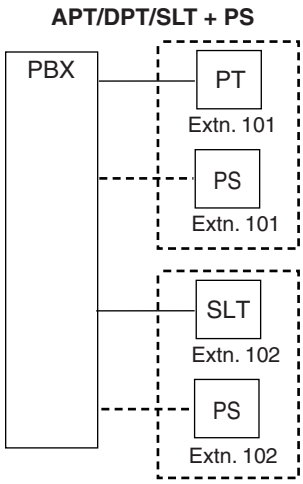
- During an Unattended Conference Call, the Unattended Conference Recall time is applied. (→ 1.13.1.2 Conference)

1.10.9 Parallelled Telephone

Description

Multiple telephones can be connected to the same port. This is useful to increase the number of telephones without additional extension cards. The combinations and features of the parallelled telephones are described below.

Features	Descriptions	Connections
Parallel Mode	Parallel connection allows the connection of an SLT to an APT or a DPT that is connected to a super hybrid port. When parallel mode is enabled, the two telephones function as follows: • Both share one extension number (main telephone's extension number). • Either telephone can make or answer a call. • If one goes off-hook while the other is on a call, the call is switched to the former.	**APT/DPT + SLT** 
EXtra Device Port (XDP) Mode	XDP mode can be used when the main telephone in a parallel connection is a DPT connected to a super hybrid port. Unlike parallel mode, each telephone can act as a completely different extension with its own extension number. (→ 2.1.1 Extension Port Configuration)	**DPT + SLT** 
Digital XDP (KX-TDA30 only)	The Digital XDP allows the connection of a DPT, to a DPT that is connected to a DPT or super hybrid port. The DPT that is connected directly to the PBX is called the "master DPT", and the DPT connected to the master is the "slave DPT". Like the XDP mode, each telephone can act as a completely different extension with its own extension number. In addition to a slave DPT, the master DPT (connected to a super hybrid port) can have an SLT in parallel or XDP mode.	**DPT + DPT**  **DPT + DPT + SLT** 

Features	Descriptions	Connections
Wireless XDP Parallel Mode	For this connection, refer to the Wireless XDP Parallel mode feature. (→ 1.23.5 Wireless XDP Parallel Mode)	

Conditions

[APT + SLT]

- If one telephone goes off-hook while the other is on a call, a three-party call is established. If one user goes on-hook, the other user continues the call.
- An extension user cannot originate a call from the SLT if the APT is:
  - playing background music (BGM)
  - receiving a paging announcement over the built-in speaker.
- The SLT will always ring for incoming calls (if its ringer is turned on). It cannot be programmed to not ring.

[DPT + SLT]

- When in the parallel mode, the XDP mode of the hybrid extension port should be off through system programming.
- Whether the SLT rings for incoming calls is programmable.

  Ring on: Both telephones ring except when the PT is in Hands-free Answerback mode (→ 1.4.1.4 Hands-free Answerback) or voice-calling mode (Alternate Receiving Ring/ Voice) (→ 1.5.3 Intercom Call).

  Ring off: Only the PT rings. However, the SLT can answer the call.
- If one goes off-hook while the other is on a call, the call is switched to the former. However, the call will not be switched in one of the following conditions:

  a) While being monitored by another extension. (→ 1.5.3 Intercom Call)

  b) While receiving OHCA (→ 1.7.4.3 Off-hook Call Announcement (OHCA)) or Whisper OHCA. (→ 1.7.4.4 Whisper OHCA)

  c) During a Conference call (→ 1.13.1 Conference Features).

  d) While Live Call Screening (LCS) or Two-way Record is activated (→ 1.22.3 Voice Mail DPT (Digital) Integration).

[DPT + DPT]

- **Available Telephones**
  KX-T7600 series (except KX-T7640)
- The PBX supports a maximum of 24 DPTs including master and slave DPTs.
- When using Digital XDP connection, the following features cannot be used with the DPTs:
  a) OHCA: A call waiting tone will be heard even if the OHCA feature is set.
  b) USB Module: The DPTs will not work correctly if a USB Module is connected. Do not connect USB Modules to the DPTs.

Note

Even if the slave DPT is disconnected, the OHCA feature and USB Module still cannot be used with the master DPT. To use them, the DPT must be disconnected from the PBX, and then reconnected.

[DPT + DPT + SLT]

- When an SLT is connected to the slave DPT in parallel mode, the SLT works as the parallel extension of the master DPT.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>
2.7.2 Parallel Connection of the Extensions
2.7.3 Extra Device Port (XDP) Connection
<KX-TDA30>
6.6.2 Parallel Connection of the Extensions
6.6.3 Extra Device Port (XDP) Connection
6.6.4 Digital Extra Device Port (Digital XDP) Connection

User Manual References

User Manual

1.7.11 Setting the Parallelled Telephone to Ring (Parallelled Telephone)

1.10.10 Calling Party Control (CPC) Signal Detection

Description

The Calling Party Control (CPC) signal is an on-hook indication (disconnect signal) sent from the analogue trunk when the other party is hung up. To maintain efficient utilisation of trunks, the PBX monitors their state and when CPC signal is detected from a line, the PBX disconnects the line and alerts the extension with reorder tone.

Conditions

- CPC signal detection is programmable for incoming trunk calls (→ LCOT CPC Signal Detection Time—Incoming [414]), and for outgoing trunk calls (→ LCOT CPC Signal Detection Time—Outgoing [413]).
- If your telephone company sends signals similar to CPC, it is recommended not to enable CPC signal detection on outgoing trunk calls.
- If CPC signal is detected during a Conference call (→ 1.13.1.2 Conference), the line is disconnected. The remaining parties stay connected.
- If CPC signal is detected during a call between a caller using the DISA feature (→ 1.15.6 Direct Inward System Access (DISA)) and an extension or an outside party, the line is disconnected.

1.11 Transferring Features

1.11.1 Call Transfer

Description

An extension user can transfer a call to another extension or an outside party. The following features are available:

Feature	Transferring method
With Announcement	Transfer is completed after announcing the destination party.
Without Announcement	Transfer is completed without an announcement. After dialling the destination, while hearing a ringback tone, the originator can replace the handset.

Call Transfer with Announcement is also known as Call Transfer—Screened.

Call Transfer without Announcement is also known as Call Transfer—Unscreened.

Conditions

- If Music on Hold is enabled, music can be sent to the held party while the call is transferred. (→ 1.12.4 Music on Hold) It is programmable whether a ringback tone or music is sent (→ Music for Transfer [712]).
- If the transfer destination extension has set FWD to an outside party, the call will be transferred to the outside party. (→ 1.3.1.2 Call Forwarding (FWD))
- COS programming determines the extensions that are able to transfer a call to an outside party (→ Call Transfer to Trunk [503]). COS can also prohibit transferring to an extension of another PBX via the TIE line service using the PBX Code method (Access with PBX Code) (→ 1.27.1 TIE Line Service).
- **Transfer Recall for Call Transfer without Announcement**
  If the transfer destination does not answer within the preprogrammed Transfer Recall time (→ Transfer Recall Time [201]), the call will be redirected to the Transfer Recall destination assigned to the extension which transferred the call. If the destination is not assigned, the call will return to the extension.
  [Available destination]

Destination	Availability
Wired Extension (PT/SLT/ISDN Extension/T1-OPX)	✔
PS	✔
Incoming Call Distribution Group	✔
PS Ring Group	
VM Group (DTMF/DPT)	
External Pager (TAFAS)	

Destination	Availability
DISA	
Analogue/ISDN Remote Maintenance	
Idle Line Access no. + Phone no.	
Trunk Group Access no. + Trunk Group no. + Phone no.	
Other PBX Extension (TIE with no PBX Code)	
Other PBX Extension (TIE with PBX Code)	

- **One-touch Transfer**
  The DSS Console and the PT user can hold a call and quickly transfer it to an extension by pressing the DSS button directly. A One-touch Dialling button can also be used if the TRANSFER command is stored with a number (→ 1.6.1 Memory Dialling Features).

User Manual References

User Manual

1.4.1 Transferring a Call (Call Transfer)

1.12 Holding Features

1.12.1 Call Hold

Description

An extension user can put a call on hold. The following features are available depending on the result.

Feature	Description
Call Hold General	Any extension can retrieve a held call.
Exclusive Call Hold	Only the extension user who held the call can retrieve it.

The result of the holding operation can be determined through system programming. Pressing the HOLD button again just after the first time alternates the mode between General and Exclusive Call Hold.

Conditions

- A PT user can hold an intercom call and multiple trunk calls simultaneously. An SLT user can hold either an intercom call or a trunk call. To hold multiple intercom calls on a PT simultaneously, use the Call Park feature (→ 1.12.2 Call Park). Call Park feature can also be used to hold multiple intercom calls and/or trunk calls simultaneously on an SLT.
- **Music on Hold**
  Music, if available, is sent to the held party. (→ 1.12.4 Music on Hold)
- **Hold Recall**
  If a call on hold is not retrieved within a preprogrammed time period (→ Hold Recall Time [200]), Hold Recall is heard at the extension which put the call on hold. If the extension is engaged in a call, the Hold Alarm will be heard.
- If an outside party is placed on hold and not retrieved within a preprogrammed time period, the call is automatically disconnected. This timer starts when Hold Recall activates.
- **Automatic Call Hold**
  A PT user can be programmed holding of the current call when pressing another CO/ICD Group/INTERCOM button, through system programming. If this feature is not enabled, the current call will be disconnected.
  [Example]
  It is possible to receive a call by pressing the flashing ICD Group button, this puts the current intercom call (on the INTERCOM button) on hold. To return to the held call, press the INTERCOM button.
- **Call Hold Retrieve Deny**
  If an extension user cannot call certain extensions on a COS basis (→ 1.1.2.2 Internal Call Block), he cannot retrieve the held call which the extensions made.
- **SLT Hold Mode**
  It is possible to choose how to hold a line and transfer a call with an SLT in the following methods through system programming:

	Hold	Transfer to Trunk	Transfer to Extension
Mode 1	Flashing the hookswitch + Going on-hook	Flashing the hookswitch + Trunk Access No.	Flashing the hookswitch + Extension No.
Mode 2 (Default)	Flashing the hookswitch + Hold Feature No. + Going on-hook	Flashing the hookswitch + Trunk Access No.	Flashing the hookswitch + Extension No.
Mode 3	Flashing the hookswitch + Hold Feature No. + Going on-hook	Flashing the hookswitch + Hold Feature No. + Trunk Access No.	Flashing the hookswitch + Extension No.
Mode 4	Flashing the hookswitch + Hold Feature No. + Going on-hook	Flashing the hookswitch + Hold Feature No. + Trunk Access No.	Flashing the hookswitch + Hold Feature No. + Extension No.

If the following occurs frequently with an SLT, choose "`Mode 2`", "`Mode 3`", or "`Mode 4`":

a) When an SLT user receives a call, reorder tone is heard or nobody answers the call.

b) When an SLT user goes off-hook, reorder tone is heard instead of a dial tone.

If a call is not terminated after going on-hook, the above cases occur. To avoid these problems, choose "`Mode 2`", "`Mode 3`", or "`Mode 4`". Every call will be terminated unless the Hold feature number is entered after flashing the hookswitch in Mode 2, Mode 3, and Mode 4.

- Hold Alarm tone pattern has a default. (→ 4.2.1 Tones/Ring Tones).

User Manual References

User Manual

1.4.2 Holding a Call

1.12.2 Call Park

Description

An extension user can place a call into a common parking zone of the PBX. The Call Park feature can be used as a transferring feature; this releases the user from the parked call to perform other operations. The parked call can be retrieved by any extension user.

Conditions

- **Automatic Call Park**
  It is possible to select an idle parking zone automatically.
- **Retry**
  If the specified parking zone is occupied or there is no vacant zone for Automatic Call Park, the originator will hear busy tone. Retrying is possible while hearing busy tone by selecting parking zone or a vacant zone.
- **Call Park Recall**
  If a parked call is not retrieved within a preprogrammed time period, Call Park Recall will be heard at the Transfer Recall destination assigned to the extension which parked the call. If the destination is engaged in a call, the Hold Alarm will be heard.
- If a parked trunk call is not retrieved within a preprogrammed time period (Default: 30 minutes), it is automatically disconnected.
- **Call Park Button**
  Pressing the Call Park button parks or retrieves a call in a preset parking zone.
  A flexible button can be customised as the Call Park button. It shows the current status of the preset parking zone as follows:

Light pattern	Status
Slow red flashing	Parked in the preset parking zone
Off	No parked call

- **Call Park (Automatic Park Zone) Button**
  Pressing the Call Park (Automatic Park Zone) button parks a call in an idle parking zone automatically. A flexible button can be customised as the Call Park (Automatic Park Zone) button.
- **Call Park Retrieve Deny**
  If an extension user cannot call certain extensions on a COS basis (→ 1.1.2.2 Internal Call Block), he cannot retrieve the parked call which the extensions made.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.4.2 Holding a Call

1.12.3 Call Splitting

Description

An extension user can alternate between two other parties. Placing the current call on hold allows the user to have a conversation with the other party.

Conditions

- While having a conversation with one party, the other party is in Consultation Hold. (**Consultation Hold:** Allows an extension user to place a call on hold temporarily to perform Call Transfer, Conference, or Call Splitting.)

User Manual References

User Manual

1.4.3 Talking to Two Parties Alternately (Call Splitting)

1.12.4 Music on Hold

Description

Music can be played to any party on hold. The following music sources are available:

a) Internal music source

b) External music source

c) Cyclic tone

[KX-TDA100/KX-TDA200]

BGM number (1 or 2) or cyclic tone can be selected for the Music on Hold (→Music on Hold [711]).

BGM numbers are the same as the external music port numbers (e.g., External music port 1 is BGM1.) However BGM2 includes internal music sources 1 and 2 as well as external music port 2. It is possible to choose what music source will be used as BGM2 (→Music Source Selection for BGM (with the KX-TDA30)/BGM2 (with the KX-TDA100/KX-TDA200) [710]).

[BGM Number and the Music Source]

BGM No.	Music Source
1	External music port 1
2	External music port 2/Internal music source 1 or 2

[KX-TDA30]

BGM or cyclic tone can be selected for the Music on Hold (→Music on Hold [711]).

BGM includes internal music sources 1, 2 and external music port. It is possible to choose what music source will be used as BGM (→Music Source Selection for BGM (with the KX-TDA30)/BGM2 (with the KX-TDA100/KX-TDA200) [710]).

Conditions

- **Hardware Requirement:** User-supplied music source
- **Volume Control**
  It is possible to change the volume of an internal and/or external music source.
- It is possible to choose a different music source for each tenant. (→2.2.3 Tenant Service)

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.11.1 Connection of Peripherals

<KX-TDA30>

6.10.1 Connection of Peripherals

1.13 Conference Features

1.13.1 Conference Features

1.13.1.1 Conference Features—SUMMARY

Description

A conference call allows a conversation between three or more parties simultaneously. The following features are available to establish a conference call:

Feature	Description	Details in
Conference	During a two-party conversation, an extension user can add other parties to establish a conference call with up to eight parties.	• 1.13.1.2 Conference
Executive Busy Override	An extension user can interrupt an existing call to establish a three-party conference call.	• 1.7.2 Executive Busy Override
Privacy Release	During a conversation with an outside party on the S-CO button, a PT user can allow another extension to join the conversation.	• 1.13.1.3 Privacy Release

Conditions

- One PBX supports a maximum of 32 simultaneous calls which are engaged in conference calls. (e.g., 4 eight-party conferences, 8 three-party conferences + 2 four-party conferences, 10 three-party conferences)

1.13.1.2 Conference

Description

This PBX supports three through eight-party conference calls. During a two-party conversation, an extension user can add other parties to their conversation, thereby establishing a conference. More than four-party conferences are only possible when a PT user originates the conference.

Unattended Conference:
A PT user can leave a conference to allow other parties to continue the conversation. The user may return to the conference, if he desires.

Conditions

- **CONF (Conference) Button**
  For a PT which does not have the CONF button, a flexible button can be customised as the Conference button.
- **Unattended Conference Call Duration**
  The call duration can be restricted by the Unattended Conference Recall time. The Unattended Conference Recall time consists of the following programmable timers:
  - **a) Unattended Conference Recall Start Time:**
    When the timer expires, Unattended Conference Recall starts to be heard at the conference originator. If the originator does not return to the conference, 1st Unattended Conference Recall Continuation time will start.
  - **b) 1st Unattended Conference Recall Continuation Time:**
    When the timer expires, an alarm tone starts to be heard at parties in the Unattended Conference. If the originator does not return to the conference, 2nd Unattended Conference Recall Continuation time will start.
  - **c) 2nd Unattended Conference Recall Continuation Time:**
    When the timer expires, the Unattended Conference call is disconnected.

  If the originator returns to the conference before the Unattended Conference call is disconnected, the timer will be cancelled. If not, Unattended Conference Recall and an alarm tone will continue to be emitted until the Unattended Conference call is disconnected.
- **For Six or More Party Conference**
  The ECHO card is required. Also the Echocanceller should be enabled through system programming.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>
2.6.4 ECHO16 Card
<KX-TDA30>
6.5.3 ECHO8 Card

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.4.5 Multiple Party Conversation

1.13.1.3 Privacy Release

Description

By default, all conversations which take place on trunk, extension lines and doorphone lines are protected by privacy (**Automatic Privacy**).
Privacy Release allows a PT user to suspend Automatic Privacy for an existing trunk call on the S-CO button in order to establish a three-party call.
System programming is required to enable or disable this feature.

Conditions

- **S-CO Button**
  A flexible button can be customised as the S-CO button.
- **Privacy Release Time**
  Privacy is released for five seconds to allow the conversation to be joined.
- This feature overrides Data Line Security (→ 1.10.5 Data Line Security) and Executive Busy Override Deny (→ 1.7.2 Executive Busy Override).

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.4.5 Multiple Party Conversation

1.14 Paging Features

1.14.1 Paging

Description

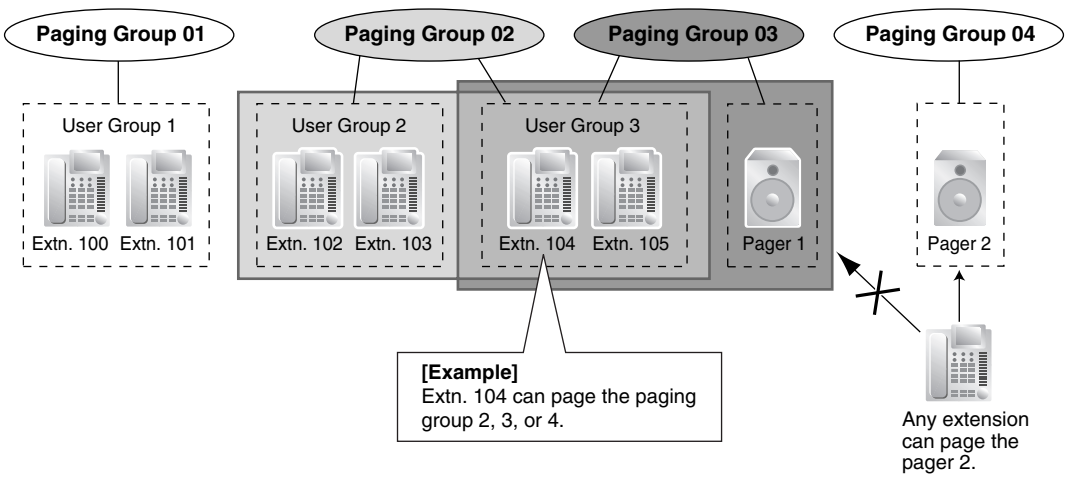
An extension user can make a voice announcement to many people simultaneously.
The message is announced over the built-in speakers of PT and/or external speakers (external pagers) which belong to the paging group. (With the KX-TDA100/KX-TDA200, up to two external pagers can be connected; with the KX-TDA30, one external pager can be connected.)
The paged person can answer the page from a nearby telephone.
It is possible to page with a call on hold in order to transfer the call.

Paging Deny:
It is possible to deny being paged on an extension basis.

Paging Group

Each paging group consists of user groups and external pagers. One user group or external pager can belong to several paging groups.
An extension can page the extension members and the external pagers of the same paging group only. If any user group is not assigned but external pager(s) is assigned in a paging group, any extension can page the external pager(s) in the paging group.
(→ 2.2.2 Group)

[Example]



[Programming Example]

Paging Group No.	User Group No.*1				External Pager No.*2	
	001	002	003	...	1	2
01	✔			...		
02		✔	✔	...		
03			✔	...	✔	
04				...		✔
05	✔	✔	✔	...	✔	✔
:	:	:	:	...	:	:

✔: Constituent
*1: → User Groups of a Paging Group [640]
*2: → External Pagers of a Paging Group [641]

Explanation:

Paging Group No.	Explanation
01	Extensions in user group 001 can page or be paged.
02	Extensions in user groups 002 and 003 can page or be paged.
03	Extensions in user group 003 can page or be paged. They can also page through the external pager 1.
04	All extensions (all user groups) can page through the external pager 2.
05	All extensions (all user groups) can page or be paged. All external pagers can also be paged.
:	:

Conditions

- Extensions which cannot be paged are:
  - PSs
  - SLTs
  - Ringing or busy PTs
  - PTs in Paging Deny mode
  - PTs in Paging DND mode

  These extensions cannot be paged but can answer the page.
- **External Pager Priority**
  External pagers can be used with the following priorities:
  TAFAS → Paging → BGM
  (→ 1.15.3 Trunk Answer From Any Station (TAFAS), 1.15.4 Background Music (BGM))

- **Volume Control**
  Paging volume from the PTs and the external pagers can be changed through system programming.
- **Paging DND**
  When DND (→ 1.3.1.3 Do Not Disturb (DND)) is set for incoming calls, it is programmable whether your extension receives paging through system programming.

User Manual References

User Manual

1.6.1 Paging

1.6.2 Answering/Denying a Paging Announcement

1.15 Optional Device Features

1.15.1 Doorphone Call

Description

A visitor using a doorphone will call a preprogrammed destination (extension or outside party). Extension users can call a doorphone.

Conditions

- **Hardware Requirement:**
  KX-TDA100/KX-TDA200: An optional Doorphone, the OPB card and DPH card
  KX-TDA30: An optional Doorphone and the DPH card
- Each doorphone port must be assigned to one tenant. The Time Table (day/lunch/break/night) of the tenant applies. (→ 2.2.4 Time Service)
- **Call Destination**
  The incoming doorphone call destination(s) can be assigned for each time mode (day/lunch/break/night) for each doorphone port (→ Doorphone Call Destination [720]). Destinations can be selected. (→ 1.1.2.1 Internal Call Features—SUMMARY)
- COS programming determines the doorphone ports that are able to make an outgoing trunk call.
- Internal Call Block determines which extensions can call a doorphone. (→ 1.1.2.2 Internal Call Block)
- **Ring Duration**
  If an incoming call is not answered within a preprogrammed time period, ringing stops and the call is cancelled.
- **Call Duration**
  The call duration can be restricted by a system timer. If the timer expires, the call will be disconnected.
- **Door Open**
  While engaged on a doorphone call, the extension user can unlock the door to let the visitor in. (→ 1.15.2 Door Open)
- A doorphone number can be referenced for each doorphone port (→ Doorphone Number Reference [729]).
- **KX-TDA30 only**
  Doorphones 1 and 2 (or 3 and 4) cannot make a call simultaneously. If a visitor presses the doorphone button while the other doorphone is engaged in a call, he will hear no tone. Doorphones 1 and 2 (or 3 and 4) cannot receive a call simultaneously. If an extension user calls a doorphone while the other doorphone is engaged in a call, he will hear a busy tone.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.6.1 OPB3 Card

2.6.2 DPH4 Card

2.6.3 DPH2 Card

2.10.1 Connection of Doorphones and Door Openers <KX-TDA30>
6.5.1 DPH4 Card
6.5.2 DPH2 Card
6.9.1 Connection of Doorphones and Door Openers

User Manual References

User Manual

1.9.1 If a Doorphone/Door Opener is Connected

1.15.2 Door Open

Description

An extension user can unlock the door for a visitor using his telephone.
The door can be unlocked by extension users who are allowed to unlock the door in the COS programming (→ Permission for Door Open Access [512]). However, while engaged on a doorphone call, any extension user can unlock the door to let the visitor in (→ 1.15.1 Doorphone Call).

Conditions

- **Hardware Requirement**: A user-supplied door opener on each door.
- The door opener will unlock the door even if a doorphone is not installed.
- **Door Open Duration**
  The door can remain unlocked for a preprogrammed time period (→ Door Open Duration Time [207]).

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.10.1 Connection of Doorphones and Door Openers

<KX-TDA30>

6.9.1 Connection of Doorphones and Door Openers

User Manual References

User Manual

1.9.1 If a Doorphone/Door Opener is Connected

1.15.3 Trunk Answer From Any Station (TAFAS)

Description

A ring tone is sent through the external pager when an incoming call is received. Any extension user can answer the call.

Conditions

- **Hardware Requirement:** A user-supplied external pager
- **Floating Extension Number**
  A floating extension number can be assigned for an external pager (→ External Pager Floating Extension Number [700]). It is possible to access an external pager by dialling their floating extension numbers.
- **Pager Volume**
  It is possible to change the volume of an external pager through system programming.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.11.1 Connection of Peripherals

<KX-TDA30>

6.10.1 Connection of Peripherals

User Manual References

User Manual

1.3.4 Answering a Call via an External Speaker (Trunk Answer From Any Station [TAFAS])

1.15.4 Background Music (BGM)

Description

A PT user can listen to BGM through the built-in speaker while on-hook and idle.

BGM—External:
BGM can also be broadcast in the office through the external pagers, this can be turned on and off by an extension assigned as the manager.

Conditions

[BGM]

- **Hardware requirement:** A user-supplied external music source, such as a radio
- The music through the PT is interrupted when going off-hook.
- **KX-TDA100/KX-TDA200 only**
  Each user can set/cancel BGM, and also select the music source.
- **KX-TDA30 only**
  Each user can only set/cancel BGM; the user cannot make a music selection.

[BGM—External]

- **Hardware requirement:** A user-supplied external pager
- External pagers can be used with the following priorities:
  TAFAS → Paging → BGM
  (→ 1.14.1 Paging, 1.15.3 Trunk Answer From Any Station (TAFAS))

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.11.1 Connection of Peripherals

<KX-TDA30>

6.10.1 Connection of Peripherals

User Manual References

User Manual

1.7.8 Turning on the Background Music (BGM)

2.1.4 Turning on the External Background Music (BGM)

1.15.5 Outgoing Message (OGM)

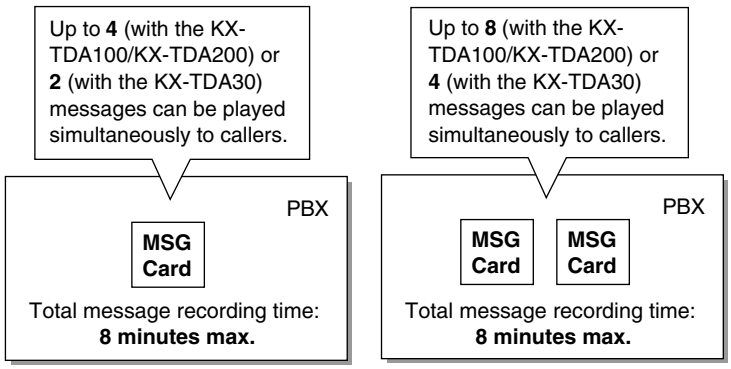
Description

An extension assigned as the manager (manager extension) can record outgoing messages (OGMs) for the following features:

Feature	Usage	Details in
Direct Inward System Access (DISA)	When a call arrives on a DISA line, the caller will hear a message.	• 1.15.6 Direct Inward System Access (DISA)
Queuing Feature	If assigned in the Queuing Time Table of the incoming call distribution group (→ Sequences in Queuing Time Table [631]), any caller who is waiting in a queue will hear a message.	• 1.2.2.3 Queuing Feature
Timed Reminder	When answering the Timed Reminder Alarm, the user will hear a message.	• 1.25.4 Timed Reminder

Conditions

- **Hardware Requirement:**
  KX-TDA100/KX-TDA200: The OPB card and MSG card
  KX-TDA30: The MSG card
- **The Number of Messages and the Recording Time**
  A maximum of 64 (with the KX-TDA100/KX-TDA200) or 32 (with the KX-TDA30) messages can be recorded. The recording time per message is a maximum of one minute. However the total recording time in the PBX is 8 minutes.
- A floating extension number can be assigned to each outgoing message (OGM) (→ Outgoing Message (OGM) Floating Extension Number [730]).
- One MSG card allows 4 (with the KX-TDA100/KX-TDA200) or 2 (with the KX-TDA30) messages to play simultaneously. The MSG card expansion does not increase the total recording time or the number of messages. It increases the number of messages that can be played simultaneously.
  [Example]


Up to **4** (with the KX-TDA100/KX-TDA200) or **2** (with the KX-TDA30) messages can be played simultaneously to callers.
PBX
MSG Card
Total message recording time:
8 minutes max.
Up to **8** (with the KX-TDA100/KX-TDA200) or **4** (with the KX-TDA30) messages can be played simultaneously to callers.
PBX
MSG Card
MSG Card
Total message recording time:
8 minutes max.

Note

When MSG card expansion has been completed, the new MSG card is blank—all necessary messages will need to be stored in the card. This can be done in two ways:

a) By installing the new MSG card then recording each message again. The PBX will automatically store the new recordings in each MSG card.

b) By transferring messages from the PBX to a personal computer (PC) using the Maintenance Console, then transferring them back to the PBX with the new MSG card installed.

- The same message can also be played simultaneously to multiple callers.
- **Recording Methods**

a) Record voice messages through the extension telephone

b) Transfer prerecorded voice messages from external sound source into the PBX via an external music port.

Note

Record voice messages only; avoid the recording of music.

- After recording messages, a manager extension can also play them back for confirmation.
- Progress tone is sent to a manager extension before recording messages during a preprogrammed time period, or during clearing the prerecorded message stored at the floating extension number of desired message. The longer one is applied.
- Record a message when traffic is light (early morning or late evening).
- When the manager tries to record a message, he will hear ringback tone if a message channel is in use. When all ports of the MSG card(s) become idle, he will hear the progress tone for a preprogrammed time period.
  After that, the PBX will automatically proceed into the recording mode.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.6.1 OPB3 Card

2.6.5 MSG4 Card

<KX-TDA30>

6.5.4 MSG2 Card

User Manual References

User Manual

2.1.5 Recording Outgoing Messages (OGM)

1.15.6 Direct Inward System Access (DISA)

Description

An outside caller can access specific PBX features as if the caller is an SLT extension user on the PBX, when the incoming call destination is a DISA floating extension number assigned to each DISA message. The caller can have direct access to features such as:

- Placing an intercom call to an extension, operator or any floating extensions (e.g., modem for remote system administration, an external pager for TAFAS).
- Calling an outside party via the PBX.
- Operating some PBX remote features (e.g., FWD)

Outgoing Message (OGM)

When a call arrives on a DISA line, a prerecorded DISA message will greet and guide the caller.
Any extension assigned as the manager can record outgoing messages (OGMs). (→ 1.15.5 Outgoing Message (OGM))

DISA Intercept—No Dial

If the caller fails to dial any digits within a preprogrammed time period (DISA 1st Dial Time for Intercept) (→ DISA Intercept Time [211]) following the outgoing message (OGM), the call is redirected to the operator extension.

DISA Built-in Automated Attendant Number (DISA AA Service)

After listening to the outgoing message (OGM), the caller may dial a single digit (DISA AA number). The destination for each DISA AA number can be assigned for each message. It is also possible to assign other DISA floating extension number as the destination (**Multistep DISA AA Service**).
If the caller dials a second digit within a preprogrammed time period (DISA 2nd Dial Time for AA), the DISA AA service is not employed.

DISA Security Mode and Available Features

If the DISA AA service is not employed, the caller may access the PBX features by entering the feature numbers. To prevent others from accessing the PBX features, it is possible to assign DISA security (→ DISA Security Mode [732]). The features available depend on the preprogrammed DISA security mode as follows:

Security Mode	Intercom Call	TIE Line Call		Trunk Call
		Without PBX Code	With PBX Code	
All Security				
Trunk Security	✔	✔		
No Security	✔	✔	✔	✔*

✔: Available

*: If trunk call is available, Account Code Entry (→ 1.5.4.3 Account Code Entry) is also available.

Note

DISA AA service and Operator Call (→ 2.2.5 Operator Features) are available for any security mode.

Security Mode Override:
If the caller operates with the following features while hearing a DISA message, the security mode can be changed to the No Security mode temporarily:

- Walking COS (→ 1.8.5 Walking COS): Walking COS feature number + extension number + extension personal identification number (PIN)
- Verified Code Entry (→ 1.8.6 Verified Code Entry): Verified Code Entry feature number + ✳ + verified code + verified code PIN

After changing mode, the mode remains in force for the duration of the call.

Available PBX Features with Security Mode Override:
After Security Mode Override is done with the Walking COS feature, the following features are available (It regards a remote operation from the assigned extension [→ 1.25.5 Remote Extension Control by User]):

- FWD
- DND
- Log-in/Log-out (→ 1.2.2.6 Log-in/Log-out)
- Absent Message (→ 1.17.2 Absent Message)
- Extension Lock (→ 1.8.3 Extension Lock)
- Time Service Switching Mode (→ 2.2.4 Time Service)

DISA Intercept Routing—Busy/DND

If the destination called by the outside party is busy or in DND mode, and if (1) the busy destination does not set Call Waiting, FWD, or Idle Extension Hunting or (2) the destination is in DND mode and Idle Extension Hunting is not available, one of the following can be selected through system programming:

a) Busy Tone: Busy tone is sent to the caller.

b) **Intercept Routing:** Intercept Routing—Busy/DND will work and the call will be redirected to the preprogrammed destination.

c) **Outgoing Message (OGM):** An outgoing message (OGM) will be sent to the caller. The message for busy/DND mode can be assigned for each outgoing message (OGM) which has a DISA floating extension number.

DISA Intercept Routing—No Answer

If a destination is not available to answer a DISA call within a preprogrammed time period (DISA Intercept time) (→ DISA Intercept Time [211]) after the call is reached, the call will be redirected to the programmed destination by the Intercept feature (→ Trunk Group Intercept Destination [470] or Extension Intercept Destination [604]).
If the intercept destination is not available to answer the call within a preprogrammed time period (DISA Disconnect Time after Intercept) after the DISA Intercept time expires, the call will be disconnected.

SMDR

The call information for DISA is recorded as the one of the DISA floating extension numbers. (→ 1.24.1 Station Message Detail Recording (SMDR))

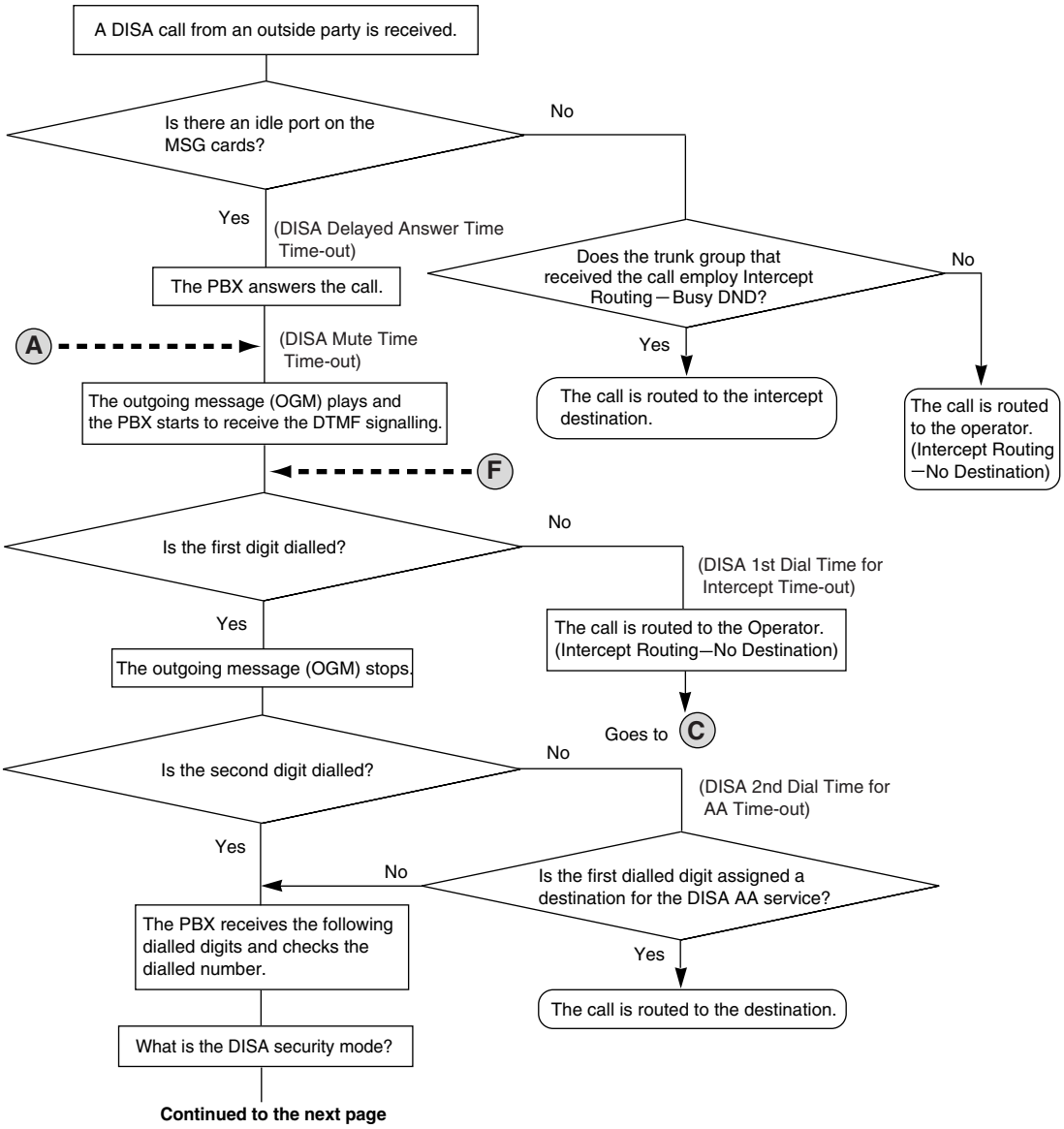
[Programming Example]

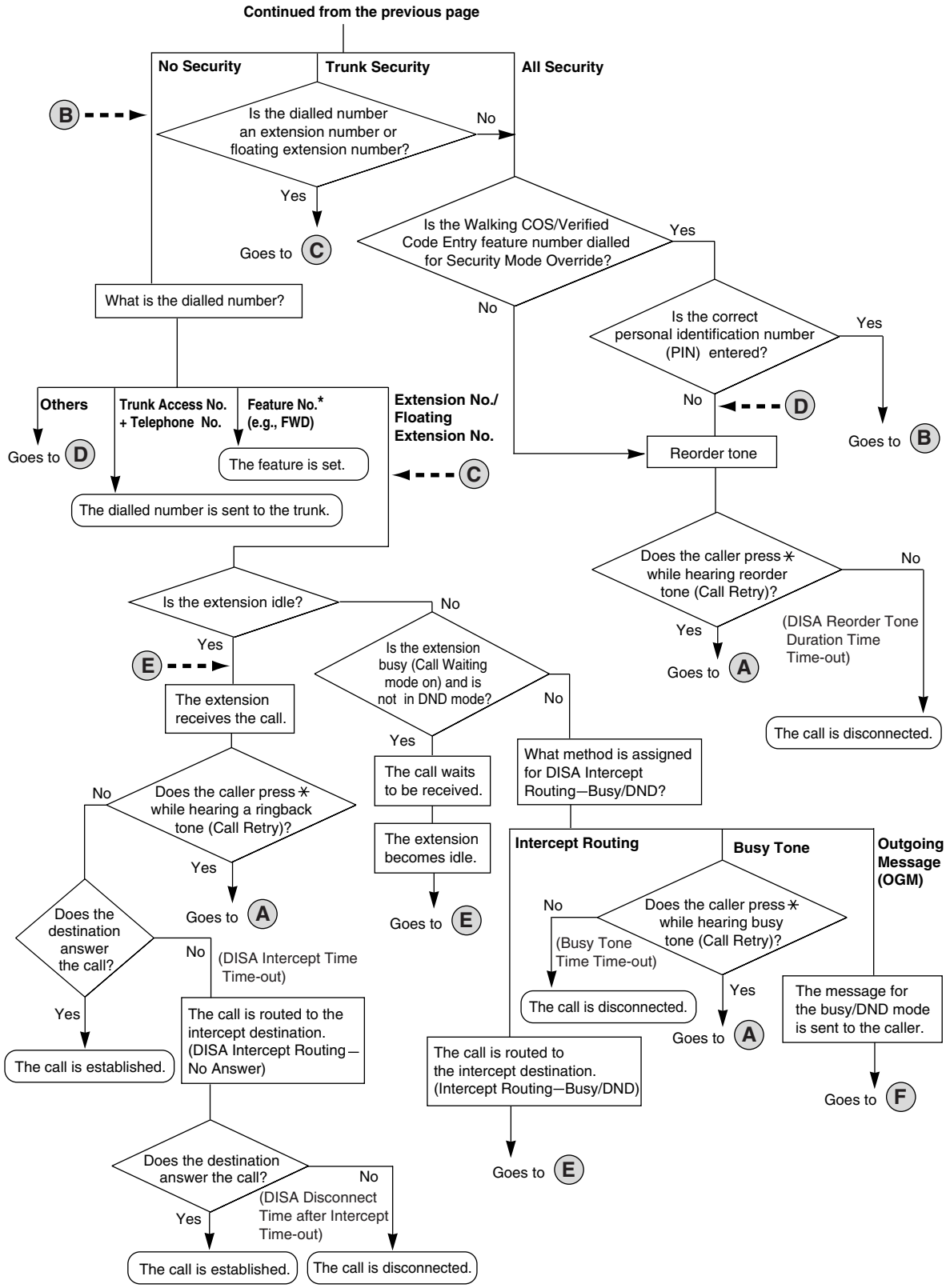
The following items can be assigned to each outgoing message (OGM):

Outgoing Message (OGM) No.	Floating Extn. No.*1	Automated Attendant No.										Busy/DND Message No.
		0	1	2	3	4	5	6	7	8	9	
01	300	100	301	200	103	202	101	102	400	104	205	04
02	301											05
:	:	:	:	:	:	:	:	:	:	:	:	:

*1: → Outgoing Message (OGM) Floating Extension Number [730]

[Flowchart]


A DISA call from an outside party is received.
Is there an idle port on the MSG cards?
No
Yes
(DISA Delayed Answer Time Time-out)
The PBX answers the call.
Does the trunk group that received the call employ Intercept Routing − Busy DND?
No
Yes
A
(DISA Mute Time Time-out)
The call is routed to the intercept destination.
The call is routed to the operator. (Intercept Routing −No Destination)
The outgoing message (OGM) plays and the PBX starts to receive the DTMF signalling.
F
Is the first digit dialled?
No
(DISA 1st Dial Time for Intercept Time-out)
Yes
The call is routed to the Operator. (Intercept Routing—No Destination)
The outgoing message (OGM) stops.
Goes to C
Is the second digit dialled?
No
(DISA 2nd Dial Time for AA Time-out)
Yes
No
Is the first dialled digit assigned a destination for the DISA AA service?
The PBX receives the following dialled digits and checks the dialled number.
Yes
The call is routed to the destination.
What is the DISA security mode?
Continued to the next page


Continued from the previous page
No Security
Trunk Security
All Security
B
Is the dialled number an extension number or floating extension number?
No
Yes
Goes to C
Is the Walking COS/Verified Code Entry feature number dialled for Security Mode Override?
Yes
No
What is the dialled number?
Is the correct personal identification number (PIN) entered?
Yes
No
D
Goes to B
Others
Trunk Access No. + Telephone No.
Feature No.* (e.g., FWD)
Extension No./ Floating Extension No.
Goes to D
The feature is set.
C
The dialled number is sent to the trunk.
Reorder tone
Does the caller press ✳ while hearing reorder tone (Call Retry)?
No
Yes
(DISA Reorder Tone Duration Time Time-out)
Goes to A
The call is disconnected.
Is the extension idle?
No
Yes
E
Is the extension busy (Call Waiting mode on) and is not in DND mode?
No
The extension receives the call.
Yes
The call waits to be received.
What method is assigned for DISA Intercept Routing—Busy/DND?
The extension becomes idle.
Goes to E
Intercept Routing
Busy Tone
Outgoing Message (OGM)
Does the caller press ✳ while hearing a ringback tone (Call Retry)?
No
Yes
Goes to A
Does the caller press ✳ while hearing busy tone (Call Retry)?
No
(Busy Tone Time Time-out)
Yes
The call is disconnected.
Goes to A
The message for the busy/DND mode is sent to the caller.
Goes to F
Does the destination answer the call?
No
(DISA Intercept Time Time-out)
Yes
The call is established.
The call is routed to the intercept destination. (DISA Intercept Routing—No Answer)
The call is routed to the intercept destination. (Intercept Routing—Busy/DND)
Goes to E
Does the destination answer the call?
No
(DISA Disconnect Time after Intercept Time-out)
Yes
The call is established.
The call is disconnected.

*: Feature numbers are available only when the Walking COS feature is used for Security Mode Override.

Conditions

WARNING

When you enable the Trunk-to-Trunk Call feature of DISA, and a third party discovers the personal identification number (PIN) (verified code PIN/extension PIN) of the PBX, there will be a risk that they will make fraudulent phone calls using the telephone line, and the cost will be charged to your own account. In order to avoid this problem, we strongly recommend the following points:

a) Carefully maintain the secrecy of the PIN.

b) Specify a complicated PIN as long and random as you can make it.

c) Change the PIN frequently.

- **Hardware Requirement:**
  KX-TDA100/KX-TDA200: The OPB card and MSG card
  KX-TDA30: The MSG card
- **DISA Delayed Answer Time**
  It is possible to set the Delayed Answer time (→ DISA Delayed Answer Time [209]) so that the caller will hear a ringback tone within a preprogrammed time period first before hearing an outgoing message (OGM).
- **DISA Mute Time**
  It is possible to set the Mute time until the outgoing message (OGM) plays and the PBX starts to receive the DTMF signalling after the caller reaches the DISA line.
- **Trunk-to-Trunk Call Duration**
  If a call between two outside parties is established, the call duration can be restricted by a system timer. (→ 1.10.8 Trunk Call Limitation) If the timer expires, the line will be disconnected unless the originating caller retries by pressing "✳" or extends the time by sending any DTMF signalling. The caller can prolong the call duration within the preprogrammed time period (→ DISA Trunk-to-Trunk Call Prolong Time [210]) and preprogramming number of times.
- **Tone Detection**
  The following three types of tone detection can be enabled for each trunk group to disconnect a trunk-to-trunk call via DISA.
  - Silence Detection (→ DISA Silence Detection [475])
  - Continuous Signal Detection (→ DISA Continuous Signal Detection [476])
  - Cyclic Signal Detection (→ DISA Cyclic Signal Detection [477])
- **Call Retry**
  While hearing a ringback, reorder, or busy tone, retrying the call is possible by pressing "✳". During a trunk-to-trunk conversation, the result of pressing "✳" is programmable: retry mode or DTMF mode.
- **DISA Reorder Tone Duration**
  It is possible to set the DISA Reorder Tone Duration time. Reorder tone will be sent to the caller for a preprogrammed time period. When the timer expires the call will be disconnected. Retrying the call is possible by pressing "✳" during the DISA Reorder Tone Duration time.
- **Call Deny**
  Extensions can deny DISA calls on a COS basis.
- **Verified Code PIN Lock/Extension PIN Lock**
  If the wrong PIN is entered incorrectly three times, the line will be disconnected. If the incorrect PIN is entered a preprogrammed number of times successively, the PIN against

the verified code or the extension will be locked. Only an extension assigned as the manager can unlock it. In this case, the PIN will be unlocked and will be cleared.

- Each outgoing message (OGM) call have a name through system programming (→ Outgoing Message (OGM) Name [731]) for programming reference.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.6.1 OPB3 Card

2.6.5 MSG4 Card

<KX-TDA30>

6.5.4 MSG2 Card

Feature Guide References

1.1.1.6 Intercept Routing

1.1.1.7 Intercept Routing—No Destination

1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)

User Manual References

User Manual

1.2.8 To Access Another Party Directly from Outside (Direct Inward System Access [DISA])

1.2.9 Setting Your Telephone from Another Extension or through DISA (Remote Setting)

1.16 Caller ID Features

1.16.1 Caller ID

Description

The PBX receives caller's information, such as the caller's name and telephone number, on trunks assigned to receive Caller ID. A display PT user can see the information.
The PBX can modify the received number according to preprogrammed tables so that he can call back or memorise it. For example, if the area code is sent and it is not required to return the call, store the area code in the table so that the area code is deleted automatically.

Caller ID to SLT Port (KX-TDA30 only):

An SLT user can also see the caller's information. This feature is available only for SLTs compatible with FSK type Caller ID.

Note

This feature complies with ETSI (European Telecommunications Standards Institute) type FSK and Bellcore type FSK.

1. **Features**
   Caller ID includes the following features:

Feature	Description	Details in
Caller ID	Caller's information which is sent from **an analogue trunk**. The following Caller ID signalling types are supported: FSK and DTMF.	–
Calling Line Identification Presentation (CLIP)	Caller's information which is sent from **an ISDN line**.	• 1.19.1.2 Calling/ Connected Line Identification Presentation (CLIP/COLP)
Automatic Number Identification (ANI) (KX-TDA100/KX-TDA200 only)	Caller's information which is sent from an **E1** or **T1 line**.	• 1.20.1 E1 Line Service (KX-TDA100/KX-TDA200 only) • 1.21.1 T1 Line Service (KX-TDA100/KX-TDA200 only)

2. Service Features

Features	Description	Details in
Calling Line Identification (CLI) Distribution	Directs a DIL/DID/DDI/MSN call to a CLI destination if the caller's identification (Caller ID/CLIP/ANI) has been assigned to the Caller ID Table.	• 1.1.1.5 Calling Line Identification (CLI) Distribution
Incoming Call Log	Caller's information is automatically recorded in the call log of the called extension. This information is used for confirming the caller, calling the caller back, and/or storing the number and name in the Personal Speed Dialling.	• 1.16.2 Incoming Call Log

3. Number/Name Assignment

3.1 Automatic Caller ID Number Modification

This PBX automatically modifies the incoming caller's number according to preprogrammed tables. The modified number will be recorded for calling back.
The PBX supports a number of modification tables, and each trunk group can select a table for use. Each table has ten locations for local/international calls and one for long distance calls.
The PBX checks the local/international call data first. If a match is not found, the long distance call data is applied.
[Example]

<Table Selection>

Trunk Group No.	Modification Table
1	1
2	3
:	:

<Modification Table>

Modification Table 1			
	Area Code	Removed No. of Digits	Added No.
Local/International Call Data 1	012	3	Blank
Local/International Call Data 2	00	2	001
:	:	:	:
Local/International Call Data 10			
Long Distance Call Data 1	Not programmable	0	0

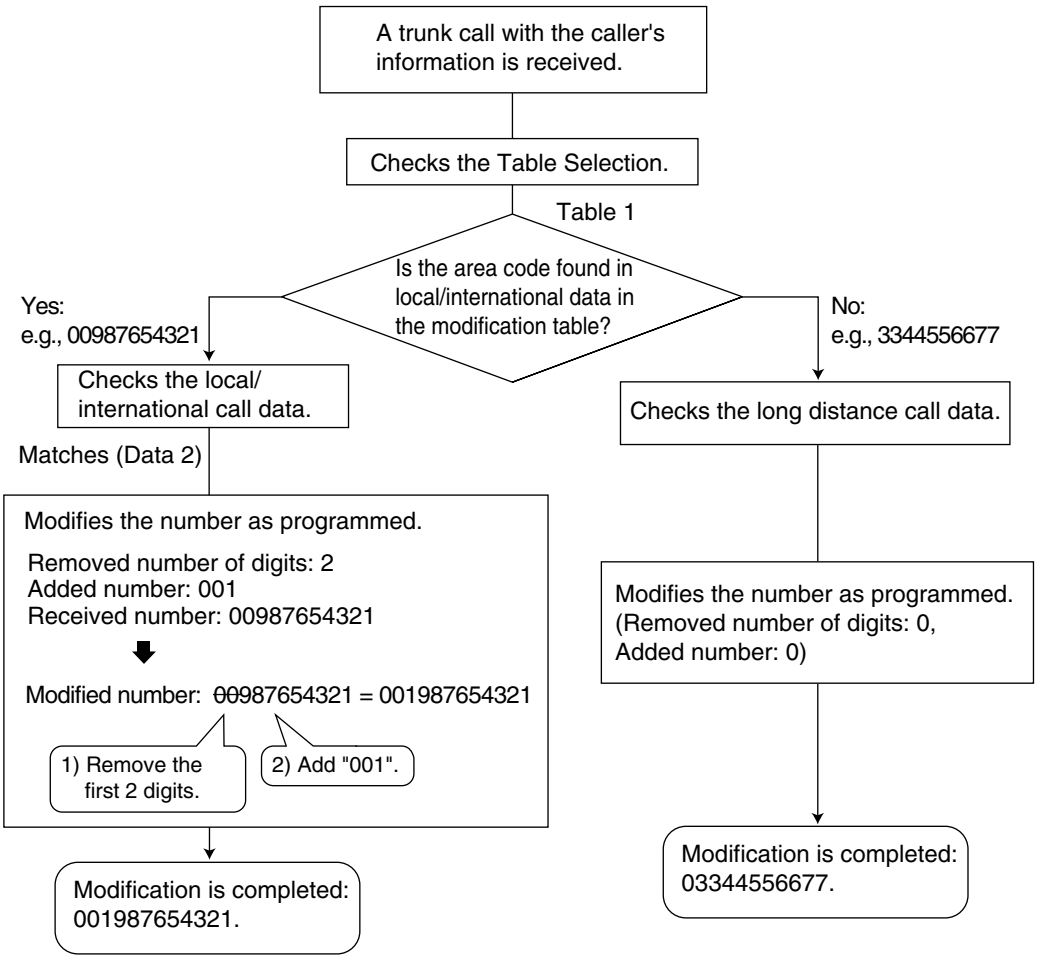
Note

When caller's information is sent through an ISDN line and the call type is Subscriber, National, or International, the following modification table is used instead of the above table:

<Modification Table>

	Removed No. of Digits	Added No.
Subscriber Call Data	0	Blank
National Call Data	0	0
International Call Data	0	00

<Modification Flowchart>


A trunk call with the caller's information is received.
Checks the Table Selection.
Table 1
Is the area code found in local/international data in the modification table?
Yes:
e.g., 00987654321
No:
e.g., 3344556677
Checks the local/ international call data.
Checks the long distance call data.
Matches (Data 2)
Modifies the number as programmed.
Removed number of digits: 2
Added number: 001
Received number: 00987654321
Modified number: 00987654321 = 001987654321
1) Remove the first 2 digits.
2) Add "001".
Modifies the number as programmed. (Removed number of digits: 0, Added number: 0)
Modification is completed: 001987654321.
Modification is completed: 03344556677.

3.2 Caller ID Table Assignment

The System Speed Dialling Table is also used as the Caller ID Table.
In each location of the table, the following items can be assigned:

a) Telephone number (Trunk Access number + caller's telephone number)

b) System Speed Dialling name (caller's name)
(shown on the display or SMDR)

c) CLI destination
(used for CLI feature)

When a caller's modified telephone number matches a telephone number (the Trunk Access number is disregarded.) in the table, the call is sent to the assigned CLI destination.

[Example]

Location (System Speed Dialling No.)	Telephone No.*1	System Speed Dialling Name.*2	CLI Destination
000	90123456789	ABC Company	200
001	:	:	:
:	:	:	:

*1: → System Speed Dialling Number [001]
*2: → System Speed Dialling Name [002]

3.3 Caller's Name Reference

A name can also be shown on the display or SMDR. The PBX searches for the name in the following order:

1) Personal Speed Dialling data of the original called extension

2) System Speed Dialling (Caller ID) Table

3) Caller ID name received from the public line (Caller ID Name Reference)

If the name is not found, it will not be displayed.

Conditions

[General]

- Caller ID signalling type can be selected through system programming (→ Caller ID Signal Type [490]).
- The Caller ID Name Reference is only available for calls from the public network.
- **For Users in New Zealand only**
  Using the toll services of a company other than Telecom NZ.
  If the PBX is set up to use the toll services of a company other than Telecom NZ, the telephone numbers dialled from the Caller Display listings within the PBX will be directed through the toll services of the other company because the telephone numbers include the toll access digit and area code digit. A toll charge may be incurred. Please check with the toll carrier concerned.

[Caller ID to SLT Port (KX-TDA30 only)]

- **Hardware Requirement:** The EXT-CID card
- Up to 8 SLTs can be programmed to receive caller's information through system programming.
- When the caller's number is sent to an SLT, a Trunk Access number can be automatically added to the telephone number through system programming for calling back.
- When the caller's number exceeds 16 digits, the SLT receives only the first 16 digits, not counting the preceding Trunk Access number (if it is programmed to be added).
- If a call is transferred to an SLT, the transferring extension's information will be shown on the SLT. If the transferring extension goes on-hook, the original caller's information will be shown.
- When the Caller ID has information, such as private, out of area, or long distance, the information will be shown instead of the caller's number and name.

- For an SLT that is connected to a PT in parallel, Caller ID information is available only when the SLT is set to "ring".
- Even if the caller's name is sent, the name may not be shown depending on the type of SLT.
- Incoming Call Log information is not shown on the SLT.

Feature Guide References

1.6.1.5 Speed Dialling—Personal/System

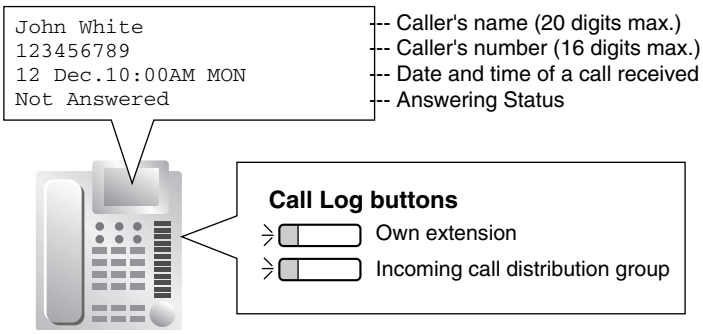
1.24.1 Station Message Detail Recording (SMDR)

1.16.2 Incoming Call Log

Description

When an incoming public trunk call with the caller's information (e.g., Caller ID) is directed to an extension, the information is automatically recorded in the call log of the called extension. This information is shown on the telephone display and is used for confirming the caller, calling back, or storing the number and name in the Personal Speed Dialling.

[Example]



Conditions

- **Call Log Button**
  A flexible button can be customised as the Call Log button for the extension or an incoming call distribution group. The button light shows the current status as follows:

Light pattern	Status
Red on	There is unchecked information.
Off	All information has been checked.

- **If the answering destination is not the original extension (FWD—No Answer and Call Pickup):**
  If a call is forwarded because of no answer or another extension picks up the call, the information is logged in the call logs of both the original destination and the answering destination.
- **Call Log for Incoming Call Distribution Group Calls**
  If a call for an incoming call distribution group is not answered, the information is logged in the call log for the incoming call distribution group. If it is answered, the information is logged in the call log for the answering extension.
- **Call Log for PS Calls**
  If a PS or a CS is in one of the following situations when a call arrives, the information is logged in the call log for the PS:
  a) When the PS is out of range.
  b) When the PS is turned off.
  c) When the CS is busy.
- **Incoming Call Log Display Lock**
  An extension user can lock the Incoming Call Log display to prevent other users from referring to the call information at his extension through personal programming (Directory and Call Log Lock). In this case, the Outgoing Call Log display and the Personal Speed

Dialling number display is also locked. An extension personal identification number (PIN) is required to use this feature. (→ 1.25.1 Extension Personal Identification Number (PIN))

- **Storing the Call Log Information in Personal Speed Dialling**
  When storing the number and name into Personal Speed Dialling from the call log information, the Idle Line Access number is automatically attached to the telephone number.
- **Incoming Call Log Memory**
  The total memory for the Incoming Call Log is determined in the PBX. The maximum number that can be logged for each extension and incoming call distribution group is also determined through system programming. If the memory becomes full, the new call record overwrites the oldest one.

Feature Guide References

1.6.1.5 Speed Dialling—Personal/System

1.18.2 Flexible Buttons

User Manual References

User Manual

1.11.1 Using the Call Log

3.1.2 Settings on the Programming Mode

1.17 Message Features

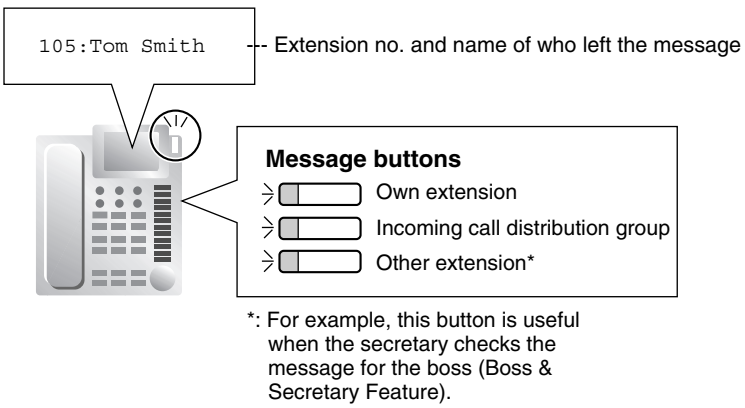
1.17.1 Message Waiting

Description

An extension user can notify another extension user that he wishes to talk to the user. The notified extension user can return the call or listen to the messages left in his mailbox of the VPS.

When a message is left on a PT, the Message button on it lights or the Message/Ringer Lamp on it turns on red. Pressing the lit Message button while on-hook shows the caller's information as shown below:

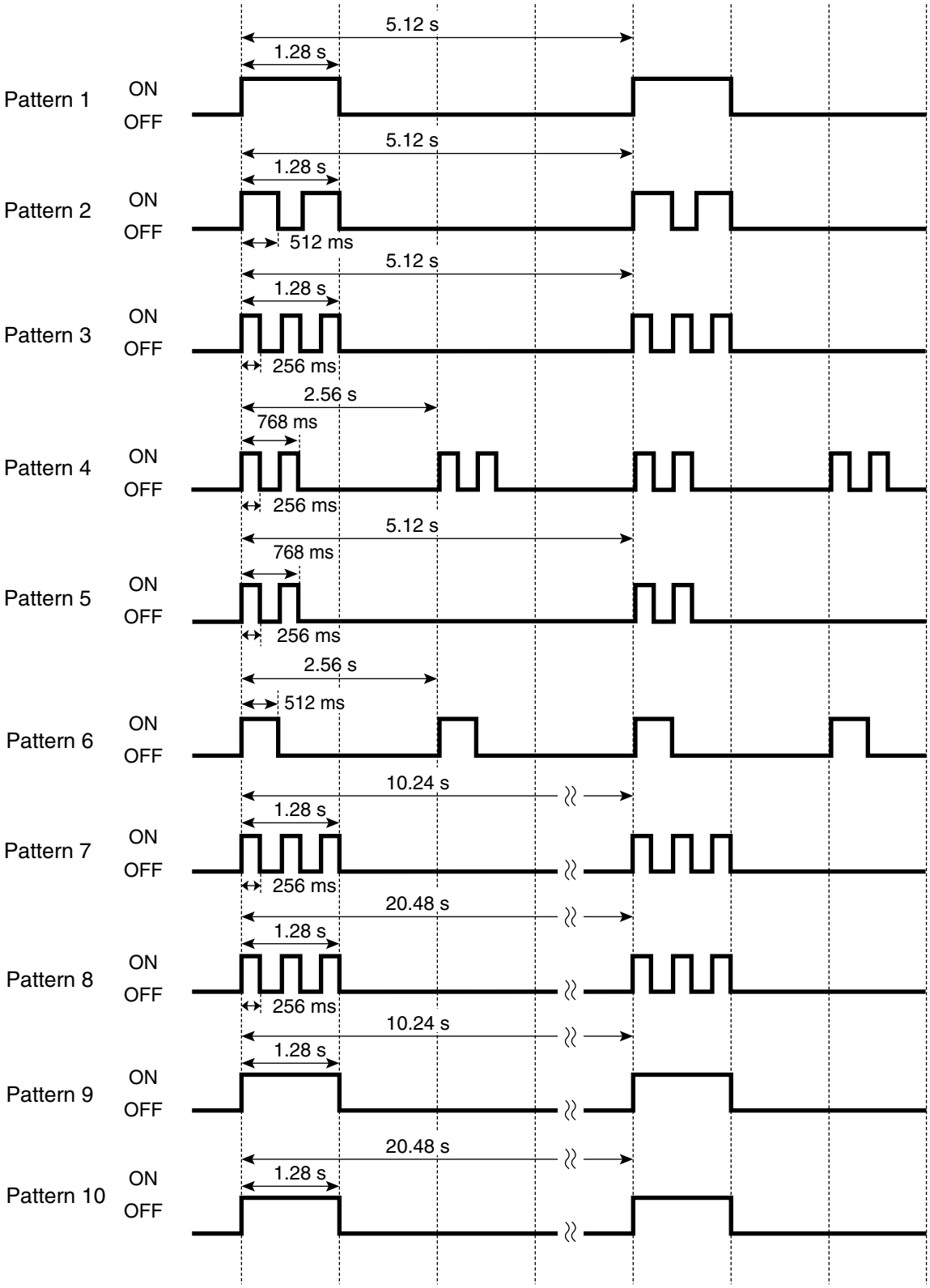
[Example]

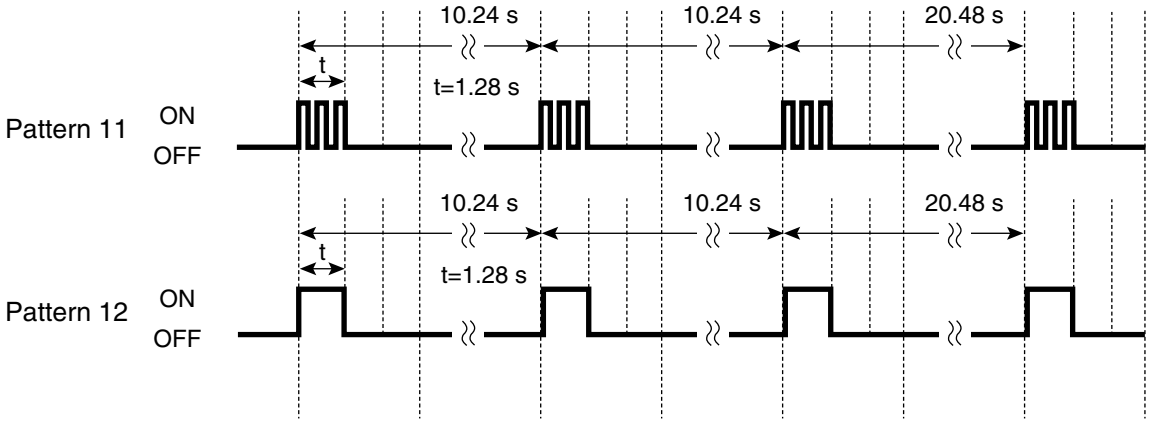

105:Tom Smith
Extension no. and name of who left the message
Message buttons
Own extension
Incoming call distribution group
Other extension*
*: For example, this button is useful when the secretary checks the message for the boss (Boss & Secretary Feature).

Conditions

- **Message Button**
  A flexible button can be customised as the Message button for the extension, other extensions, or an incoming call distribution group.
- **Distinctive Dial Tone for Message Waiting**
  If the Distinctive Dial mode is enabled, dial tone 4 will be sent to an extension when a message has been left on the extension. (→ 1.26.1 Dial Tone)
- **SLT with a Message Waiting Lamp (KX-TDA100/KX-TDA200 only)**
  The lamp activates in the same way as the MESSAGE button on a PT, if an MSLC card is connected to the PBX. The Message Waiting Lamp light pattern can be selected from the twelve patterns shown below. System programming for each extension port is required to use this feature.

[SLT Message Waiting Lamp Light Patterns]





- It is possible to set Message Waiting while hearing a ringback tone, busy tone, or DND tone.
- Messages are always left on the original extension. Messages cannot be sent to their FWD destination (→ 1.3.1.2 Call Forwarding (FWD)).
- Both the message sending extension and receiving extension can cancel a left message.
- If the message receiving extension calls the sending extension and is answered, the message will be cleared automatically. However, if the message has been left on the VPS, it depends on the VPS.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.2.4 When the Dialled Line is Busy or There is No Answer

1.9.3 If a Voice Processing System is Connected

1.17.2 Absent Message

Description

The extension users can set the message (e.g., the reason for absence, where he is) on his telephone. When display PT user calls the extension, the message is shown on the caller's telephone.

The following messages can be programmed as desired:

Type	Message No.	Message (Example)	Description
System message	1	`Will Return Soon`	Messages may be edited through system programming (→ Absent Message [008]). They are used for every extension user commonly.
	2	`Gone Home`	
	3	`At Ext %%%%` (Extension Number)	
	4	`Back at %%:%%` (Hour:Minute)	
	5	`Out until %%/%%` (Month/Day)	
	6	`In a Meeting`	
	7		
	8		
Personal message	9		A message is programmable at each extension through personal programming (Personal Absent Message), which can only be used by that extension user.

Note

The "%" means a parameter to be entered when assigning a message at an individual extension.
Up to seven "%"s can be stored for each message.

Conditions

- An extension user can select only one message at a time. The selected message is displayed at the extension while on-hook.

User Manual References

User Manual

1.5.2 Showing a Message on the Caller's Telephone Display (Absent Message)

3.1.2 Settings on the Programming Mode

1.18 Proprietary Telephone (PT) Features

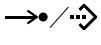
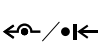
1.18.1 Fixed Buttons

Description

PTs, DSS Consoles, and Add-on Key Modules are provided with the following feature/Line Access buttons:
Depending on your device type, some buttons may not be provided.
As for buttons on PSs, please refer to the Operating Instructions for each PS.

[PT and Add-on Key Module]

Button		Usage
Navigator Key, Jog Dial, Volume Key		Used to adjust the ringer, speaker, handset and headset volume and the display contrast. Navigator Key and Jog Dial can also be used to select data from the Call Directory and the System Feature Access Menu on the display.
ENTER		Used to confirm the selected item.
CANCEL		Used to cancel the selected item.
PROGRAM	*PROG.*	Used to enter and exit the programming mode.
FLASH/ RECALL	R	Used to disconnect the current call and make another call without hanging up (Flash/Recall mode or Terminate mode) or used to send a flash/recall signal to the telephone company or host PBX to access their features (External Feature Access mode). With the KX-TDA30, also used as a CANCEL button while on-hook.
HOLD		Used to place a call on hold.

Button		Usage
SP-PHONE (Speakerphone)		Used for hands-free operation. Also used to switch between handset and hands-free operation.
MONITOR		Used for a hands-free dialling. Also used to monitor the party's voice in hands-free mode.
MESSAGE		Used to leave a message waiting indication or call back the party who left the message waiting indication.
REDIAL		Used to redial the last dialled number.
TRANSFER		Used to transfer a call to another party.
Flexible CO (Trunk)		Used to make or receive a trunk call or can be reassigned to a different Trunk Access button (Default: S-CO) or to another feature button.
INTERCOM	*INT'*	Used to make or receive intercom calls.
AUTO ANS (Auto Answer)/ MUTE		Used to receive an incoming call in hands-free mode, or used for microphone or handset mute during a conversation. (Dual feature button)
VOICE CALL/ MUTE		Used to monitor an intercom call automatically (a hands-free conversation is not possible). Also used for handset microphone mute during a conversation.
AUTO DIAL/ STORE		Used for System/Personal Speed Dialling and storing programme changes.
CONF (Conference)		Used to establish a multiple-party conversation.
FWD/DND		Used to perform FWD or DND. (Dual feature button)
PAUSE		Used to insert a pause in a stored number. With an APT, it is used as the PROGRAM button.
Soft		Used to select the item displayed on the bottom line of the display.
SELECT		Used to select the displayed item or to call the displayed number.
SHIFT		Used to access the second level of Soft button items.
MODE		Used to shift the display to access various features.

[DSS Console]

Button		Usage
ANSWER		Used to answer an incoming call or place the current call on hold and answer another call with one-touch.
RELEASE		Used to disconnect the line during or after a conversation or to complete a Call Transfer.

Button	Usage
Flexible CO (Trunk)	Used to make or receive a trunk call or can be reassigned to a different trunk or to another feature button.
Flexible DSS (Direct Station Selection)	Used to access an extension with one-touch. Every button is programmed to correspond to an extension. DSS buttons can also be reassigned to other features.
PF (Programmable Feature)	Used to access a preprogrammed feature with one-touch. (no default)

Conditions

- Certain buttons are equipped with a light to show line or feature status.

1.18.2 Flexible Buttons

Description

Flexible buttons are customised by either system or personal programming. The following types of flexible buttons are provided on PTs, DSS Consoles, Add-on Key Modules, and/or PSs:

a) Flexible CO buttons

b) Flexible Direct Station Selection (DSS) buttons

c) Programmable Feature (PF) buttons

[Button Usage]

Button	Usage
Single-CO (S-CO)	Used to access a specified trunk for making or receiving calls.
Group-CO (G-CO)	Used to access an idle trunk in a specified trunk group for making calls. Incoming calls from trunks in the assigned trunk group arrive at this button.
Loop-CO (L-CO)	Used to access an idle trunk for making calls. Incoming calls from any trunk arrive at this button.
Direct Station Selection (DSS)	Used to access an extension with one-touch.
One-touch Dialling	Used to access a preprogrammed party or feature with one-touch.
Incoming Call Distribution (ICD) Group	Used to access a specified incoming call distribution group for making or receiving calls.
Message	Used to leave a message waiting indication or call back the party who left the message waiting indication.
FWD/DND (External/ Internal/Both) *	Used to perform the FWD or DND feature for the extension. The feature is applied to trunk calls, intercom calls, or both.
Group FWD (External/ Internal/Both)	Used to perform the FWD feature for a specified incoming call distribution group. The feature is applied to trunk calls, intercom calls, or both.
Account Code Entry (Account)	Used to enter an account code.
Conference	Used to establish a multiparty conversation.
Terminate	Used to disconnect the current call and make another call without hanging up.
External Feature Access (EFA)	Used to send a flash/recall signal to the telephone company or host PBX to access their features.
Charge Reference	Used to check the total call charge for your own extension.
Call Park	Used to park or retrieve a call in a preset PBX parking zone.
Call Park (Automatic Park Zone)	Used to park a call in an idle PBX parking zone automatically.

Button	Usage
Call Log	Used to show the incoming call information.
Log-in/Log-out *	Used to switch between the log-in and log-out mode.
Hurry-up	Used to redirect the longest waiting call in the queue of an incoming call distribution group to the overflow destination.
Wrap-up *	Used to switch the Wrap-up/Not Ready and Ready modes.
System Alarm	Used to confirm a PBX error.
Time Service *	Used to switch the assigned time modes: day, lunch, break or night. Also used to check the current time mode status.
Answer	Used to answer an incoming call.
Release	Used to disconnect the line during or after a conversation, or to complete a Call Transfer.
Toll Restriction/Call Barring	Used to change the TRS/Barring level of other extension users temporarily.
ISDN Service	Used to access an ISDN service.
Calling Line Identification Restriction (CLIR) *	Used to switch between the CLIP and CLIR service.
Connected Line Identification Restriction (COLR) *	Used to switch between the COLP and COLR service.
ISDN Hold	Used to transfer a call using the telephone company.
Headset	Used to turn on/off the headset mode while idle. Used to switch between hands-free mode and headset modes during a conversation.
Time Service Switching Mode (Automatic/Manual) *	Used to switch between the Automatic Switching and Manual Switching mode.
Two-way Record	Used to record a conversation into your own mailbox.
Two-way Transfer	Used to record a conversation into the mailbox of a specified extension.
One-touch Two-way Transfer	Used to record a conversation into the mailbox of a specified extension with one-touch.
Live Call Screening (LCS)	Used to monitor your own voice mailbox while an incoming caller is leaving a message and, if desired, intercept the call.
Voice Mail (VM) Transfer	Used to transfer a call to the mailbox of a specified extension.

*: **One-touch Feature Setting Buttons:** Pressing these buttons while on-hook changes the feature settings. The new mode will be displayed for a preprogrammed time period.

Conditions

- The Flexible DSS button lights of the KX-T7040 and KX-T7240 shows only "Off" or "Red On". If the button is customised as a button which shows other light patterns, the other light patterns also show only as "Red On".

User Manual References

User Manual

3.1.3 Customising the Buttons

1.18.3 LED Indication

Description

The light (LED: Light Emitting Diode) of the Message/Ringer Lamp and following buttons (Line Status Buttons and Corresponding Extension Status Button) show line conditions with a variety of light patterns.
Line Status Buttons: S-CO, G-CO, L-CO, INTERCOM, ICD Group
Corresponding Extension Status Button: DSS

1. **Light Pattern of the Message/Ringer Lamp**
   - Incoming call from a trunk: Red flashing
   - Incoming call from another extension: Green flashing
   - Message(s) present (no incoming call): Red on
   - No Message(s) present (no incoming call): Off

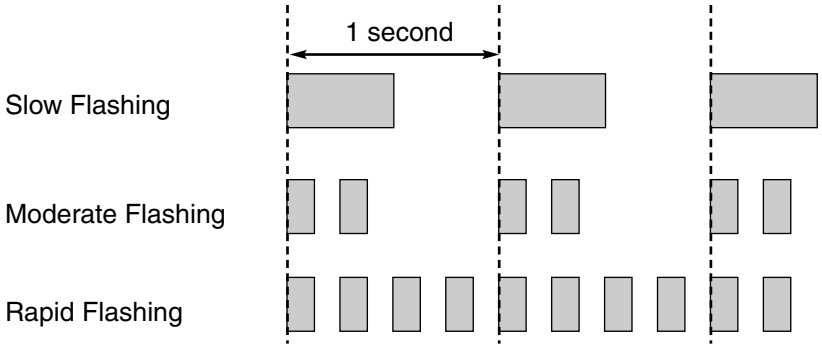
2. **Light Pattern of the Line Status Buttons**

<table>
<tr><th rowspan="2">Line Status Button: / Light Pattern</th><th colspan="3">Trunk Status</th><th>Intercom Line Status</th><th>Incoming Call Distribution Group Line Status</th></tr>
<tr><th>S-CO</th><th>G-CO</th><th>L-CO</th><th>INTERCOM</th><th>ICD Group</th></tr>
<tr><td>Off</td><td colspan="5">Idle</td></tr>
<tr><td>Green on</td><td colspan="5">This extension is using the line.</td></tr>
<tr><td>Slow green flashing</td><td colspan="5">This extension is holding the line.</td></tr>
<tr><td>Moderate green flashing</td><td colspan="5">This extension is holding the line using Exclusive Hold or using the line for an Unattended Conference.</td></tr>
<tr><td>Rapid green flashing</td><td>Incoming call/Privacy Release</td><td colspan="4">Incoming call</td></tr>
<tr><td>Red on</td><td>Incoming call for another extension/ Another extension is using the line/Another extension has the line on Exclusive Hold.</td><td>Other extensions are using all trunks in the trunk group.</td><td>–</td><td>–</td><td>This extension is logged out of the incoming call distribution group.</td></tr>
<tr><td>Slow red flashing</td><td>Another extension is holding the line.</td><td>–</td><td>–</td><td>–</td><td>–</td></tr>
<tr><td>Rapid red flashing</td><td colspan="3">Incoming call to the incoming call distribution group in Ring distribution method</td><td>–</td><td>–</td></tr>
</table>

3. Light Pattern of the Corresponding Extension Status Button

Corresponding Extension Status Button / Light Pattern	DSS
Off	Idle
Red on	Busy/DND for trunk calls
Rapid red flashing	Incoming call

4. Flashing Light Patterns



Conditions

- The incoming call shows on the buttons in the following priority:
  ICD Group→S-CO→G-CO→L-CO→INTERCOM

Feature Guide References

1.2.2.2 Group Call Distribution
1.2.2.6 Log-in/Log-out
1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)
1.12.1 Call Hold
1.13.1.2 Conference
1.13.1.3 Privacy Release

1.18.4 Display Information

Description

A display PT shows the user the following information while making or receiving calls if they are available:

Display Item	Display Example	Condition
The extension number and name of the calling or called extension, or incoming call distribution group	`123: Tom Smith`	–
Status of the called extension	`123: Busy`	–
The number and name of the optional device	`Door 02: 1st Door`	–
The dialled telephone number	`1234567890`	–

The received call information
a) Caller's name
b) Caller's number
c) Trunk number/name
d) Original Destination, if the call is forwarded
e) DDI/DID/MSN name or number
	`ABC Company`
	`12345678`
	`Line 001: Sales`
	`→102:Mike`
	`Panasonic`
		The first line message can be either (a), (c), or (e) at each extension through system programming.

Call charge fee during a trunk call.	`12.35€`	The currency (→ Currency [131]), position of the currency symbol, and the decimal point (→ Decimal Point Position for Currency [130]) are programmable.
Call duration during a trunk call.	`Line 001 11:02'28`	–

Conditions

- **Multilingual Display**
  Each extension can select its display language through personal programming (Display Language Selection).
- **Display Contrast**
  It is possible to adjust the display contrast through personal programming (Display Contrast Selection). This is available only for DPTs.
- **Display Backlight**
  Each extension can select its display backlight on or off through personal programming (Display Backlight Selection). This is available for the KX-T7633 and KX-T7636.
- Characters (name) or digits (number) exceeding the display's size limitation are not displayed. In this case, information which have been programmed is hidden, but not altered.

User Manual References

User Manual

3.1.2 Settings on the Programming Mode

1.19 Integrated Services Digital Network (ISDN) Service Features

1.19.1 Integrated Services Digital Network (ISDN)

1.19.1.1 Integrated Services Digital Network (ISDN)—SUMMARY

Description

ISDN is a digital switching and transmission network. ISDN transmits voice, data, and image in digital format. ISDN lines, if available, can be connected to public line (trunk), private line (QSIG), or ISDN terminal devices (extension).

1. ISDN Interface and Configuration

Interface Type	Description	Port Mode	Configuration Type*
Basic Rate Interface (BRI)	Provides two 64 kbps B channels for communication and one 16 kbps D channel for signalling (2B + D).	Trunk, Extension	Point-to-Point/ Point-to-multipoint
		QSIG (Master, Slave)	Point-to-Point
Primary Rate Interface (PRI)	Provides thirty or twenty-three 64 kbps B channels for communication and one 64 kbps D channel for signalling (30B + D/ 23B + D).	Trunk, QSIG (Master, Slave), Extension	Point-to-Point

*: → BRI Configuration [426] *For BRI only

Note

Point-to-Point (P-P):
One ISDN terminal device can be connected to one ISDN port.

Point-to-multipoint (P-MP):
A maximum of eight ISDN terminal devices can be connected to one ISDN port.

2. ISDN Supplementary Service Table for Public Network

Service	Description	Details in
Direct Dialling In (DDI)	Directs a call with a DDI number to a preprogrammed destination. (P-P only)	• 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)
Multiple Subscriber Number (MSN)	One ISDN port can support a maximum of 10 MSNs for routing incoming calls. (P-MP only)	• 1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service
Calling Line Identification Presentation (CLIP)	Sends the caller's telephone number to the network when making a call. The called party can see the number on his telephone display before answering the call.	• 1.19.1.2 Calling/ Connected Line Identification Presentation (CLIP/COLP)
Connected Line Identification Presentation (COLP)	Sends the telephone number of the answered party to the network when answering a call. The caller can see the number on his telephone display when the line is connected.	
Calling Line Identification Restriction (CLIR)	Prevents the caller's CLI being presented to the called party by the caller.	
Connected Line Identification Restriction (COLR)	Prevents COLP being sent by the answered party.	
Subaddressing (SUB)	You may add digits after the telephone number. These digits will be passed to ISDN terminal device.	–
User-to-user Signalling Type 1 (UUS-1)	Transmits a specified volume of the data through the D channel between the ISDN terminal devices of different PBXs using UUS types 1.	–
Advice of Charge (AOC)	The PBX can receive the call charge information on ISDN lines from the telephone company.	• 1.19.1.3 Advice of Charge (AOC)
Call Forwarding (CF)—by ISDN	Forwards an incoming call to another outside party using the ISDN service of telephone company.	• 1.19.1.4 Call Forwarding (CF)—by ISDN (P-MP) • 1.19.1.5 Call Forwarding (CF)—by ISDN (P-P)

Service	Description	Details in
Call Hold (HOLD)—by ISDN	Puts one ISDN call on hold.	• 1.19.1.6 Call Hold (HOLD)—by ISDN
Call Transfer (CT)—by ISDN	Transfers an ISDN call to an outside party. Call Transfer with Announcement and Call Transfer without Announcement are available.	• 1.19.1.7 Call Transfer (CT)—by ISDN
Malicious Call Identification (MCID)	An extension user can ask the telephone company to trace a malicious caller. Information on the malicious call will be received later on.	• 1.19.1.8 Malicious Call Identification (MCID)
Completion of Calls to Busy Subscriber (CCBS)	If an outside called party is busy, an extension can receive callback ringing when the called party becomes free.	• 1.19.1.9 Completion of Calls to Busy Subscriber (CCBS)

3. **ISDN Centrex Service**

   An extension user can have access to the features of the ISDN Centrex Service of the telephone company (e.g., Call Transfer). This is used by putting the current ISDN call on hold by sending a flash/recall signal. This feature is enabled or disabled for each ISDN port.

4. **Private Networking Connection (QSIG)**

   Private networking with QSIG is possible using an ISDN line. The QSIG mode, Master or Slave, can be enabled on an ISDN port basis.

5. **ISDN Extension** (→ 1.19.1.10 ISDN Extension)

   An ISDN (BRI/PRI) port can be used for extension connection. While the extension connection is enabled, ISDN terminal devices (e.g., ISDN telephone, G4 fax machine, personal computer) or a behind PBX can be connected to the port.
   When the ISDN port is in P-P configuration, one ISDN terminal device can be connected to the port. When the ISDN port is in P-MP configuration, a maximum of eight ISDN terminal devices can be connected to the port. However, only a maximum of two devices can be used simultaneously.

Conditions

- **Overlap/En bloc**
  ISDN call dialling mode, Overlap or En bloc, can be selected for each ISDN port. When "`Overlap`" is selected, the PBX sends each dialled digit individually. When "`En bloc`" is selected, the PBX sends all of dialled digits at once after the extension user completes dialling. The PBX recognises end of dialling as follows:
  - Dialling #, if programmed
  - Dialling a preprogrammed telephone number
  - Inter-digit timer expires.

- Some supplementary services are provided by the key protocol (→ 1.19.1.11 ISDN Service Access by Keypad Protocol).
- **Extension Number**
  An extension number can be assigned to each ISDN port.
- **Network Type**
  Each ISDN port can be set to the requirements of each country/area (→ BRI Network Type [420] *For BRI only).
- **L1 Active Mode and L2 Data Link Mode**
  The active mode of Layer 1 (L1) (→ BRI Layer 1 Active Mode [424] *For BRI only) and the data link mode of Layer 2 (L2) (→ BRI Layer 2 Active Mode [425] *For BRI only) can be selected individually, Permanent or Call by Call, on an ISDN port basis.
- **TEI Assignment Mode**
  The Terminal Endpoint Identifier (TEI) assignment mode can be selected on an ISDN port basis (→ BRI TEI Mode [427] *For BRI only).
- **Bearer Mode**
  The bearer mode can be assigned on an extension basis.
- **Automatic ISDN Configuration**
  ISDN (BRI) port configuration can be set automatically through system programming. The following items will be set by making and receiving a call using the subscriber number assigned for each ISDN (BRI) port:
  - L1 Active Mode
  - L2 Data Link Mode
  - Access Mode (Point-to-Point/Point-to-Multipoint)
  - TEI Assignment Mode (Fix 00–63/Automatic)

Feature Guide References

1.27.3 QSIG Network

1.19.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)

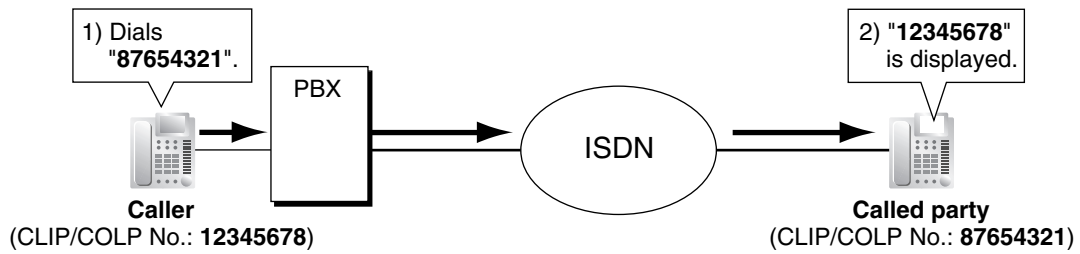
Description

Calling Line Identification Presentation (CLIP):
The PBX can send a preprogrammed telephone number to the network when an extension user makes a call. The called party can see the number on his telephone display before answering the call.

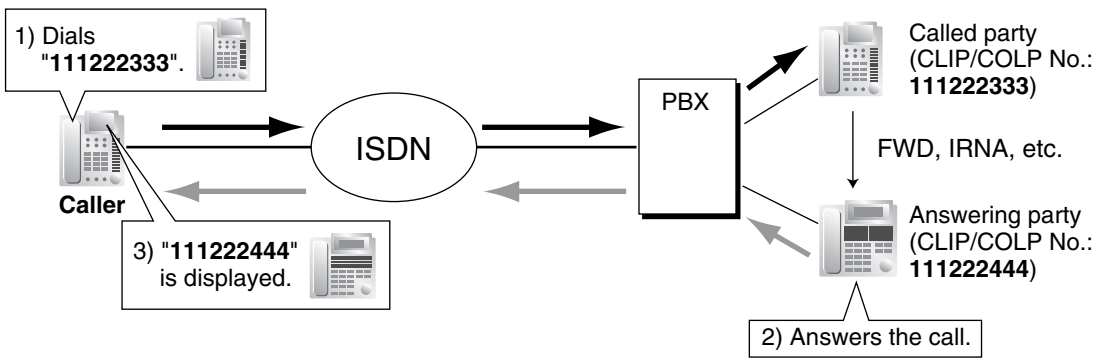
Connected Line Identification Presentation (COLP):
The PBX sends a preprogrammed telephone number to the network when the extension user answers an incoming call. The caller can see the number of the answering party on his telephone display when the call is answered.

These features comply with the following European Telecommunication Standard (ETS) specifications:
ETS 300 092 Calling Line Identification Presentation (CLIP) supplementary service.
ETS 300 097 Connected Line Identification Presentation (COLP) supplementary service.

[CLIP Example]



[COLP Example]



CLIP/COLP Number:
The telephone numbers sent to the network for CLIP/COLP can be assigned as follows:

- CLIP/COLP number for each ISDN port (subscriber's number) (→ BRI Subscriber Number [422] *For BRI only)
- CLIP/COLP number for each extension (→ CLIP/COLP Number [606])
- CLIP/COLP number for each incoming call distribution group

Each extension can select either the CLIP/COLP number for the ISDN port or the extension to be used. The CLIP/COLP number for the incoming call distribution group is used when making a call by pressing the ICD Group button or receiving a call which arrives at the ICD Group button.

Calling/Connected Line Identification Restriction (CLIR/COLR):
It is possible for each extension to restrict the sending of its telephone number to the network by pressing the CLIR button, COLR button, or entering the feature number.

These features comply with the following European Telecommunication Standard (ETS) specifications:
ETS 300 093 Calling Line Identification Restriction (CLIR) supplementary service.
ETS 300 098 Connected Line Identification Restriction (COLR) supplementary service.

Conditions

- The availability of this feature is dependent on the contract with the telephone company.
- The CLIP/COLP number for the connected ISDN port can be used for the ISDN terminal devices which cannot be assigned their own CLIP/COLP number, such as a doorphone.
- **COLP/CLIR/COLR Assignment for Each Port**
  Each service can be enabled or disabled on each ISDN port of the PBX.
- **CLIR Button and COLR Button**
  It is possible to switch between CLIP and CLIR by pressing the CLIR button, and COLP and COLR by pressing the COLR button. A flexible button can be customised as the CLIR or COLR button.
- The CLIP/COLP number must match the telephone number provided by the telephone company. Otherwise it will be ignored or replaced by another number.
- When using a private network, the extension number assigned for each extension through system programming (→ Extension Number [003]) is sent for CLIP/COLP. (→ 1.27.3.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG)

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.7.4 Displaying Your Telephone Number on the Called Party and Caller's Telephone (Calling/Connected Line Identification Presentation [CLIP/COLP])

1.7.5 Preventing Your Telephone Number Being Displayed on the Caller's Telephone (Connected Line Identification Restriction [COLR])

1.7.6 Preventing Your Number Being Displayed on the Called Party's Telephone (Calling Line Identification Restriction [CLIR])

1.19.1.3 Advice of Charge (AOC)

Description

The PBX can receive the call charge information on ISDN lines from the telephone company. There are the following types:

Type	Description
Advice of Charge During Call (AOC-D)	AOC is received during the call and when the call is completed.
Advice of Charge At End of Call (AOC-E)	AOC is received when the call is completed.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 182 Advice of Charge (AOC) supplementary service Digital Subscriber Signalling System No. One (DSS1) protocol.

Conditions

- A DPT user can see the call charge information on the display during the call.
- **Budget Management**
  If the amount of call charge reaches the preprogrammed limit, an extension user cannot make further calls. (→ 1.8.2 Budget Management)
- **AOC for ISDN extension**
  An ISDN extension also receives AOC.

1.19.1.4 Call Forwarding (CF)—by ISDN (P-MP)

Description

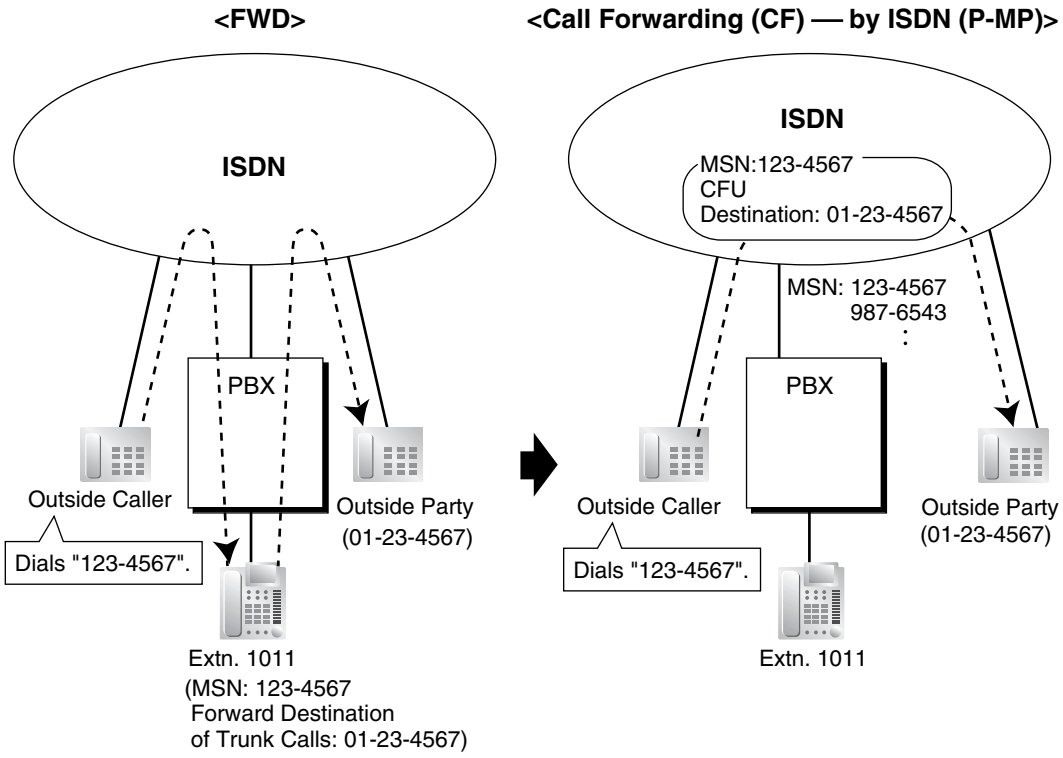
An extension user can forward the incoming ISDN call to another outside party using the ISDN service of the telephone company, instead of the PBX feature, when the call is received through an ISDN line.
Extension users can set the forward destination to the network on an MSN basis. The setting must have been done beforehand to use this feature. There are the following types:

Type	Description
Call Forwarding—Unconditional (CFU)	Incoming calls are forwarded unconditionally.
Call Forwarding—Busy (CFB)	An incoming call is forwarded when the destination on the PBX is busy.
Call Forwarding—No Reply (CFNR)	An incoming call is forwarded when the destination on the PBX does not answer within a preprogrammed time period.

When the CFB or CFNR has been set, the network determines to forward the incoming call to the preprogrammed destination after the call reached the PBX once. When the CFU has been set, the network forwards the call directly to the preprogrammed destination.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 207 Diversion supplementary service.

[Example]



Conditions

- The availability of this feature is dependent on the contract with the telephone company.
- The feature requires the MSN service. (→ 1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service)
- COS programming determines the extensions that are able to use this feature.

Feature Guide References

1.3.1.2 Call Forwarding (FWD)

User Manual References

User Manual

1.5.1 Forwarding Calls

1.19.1.5 Call Forwarding (CF)—by ISDN (P-P)

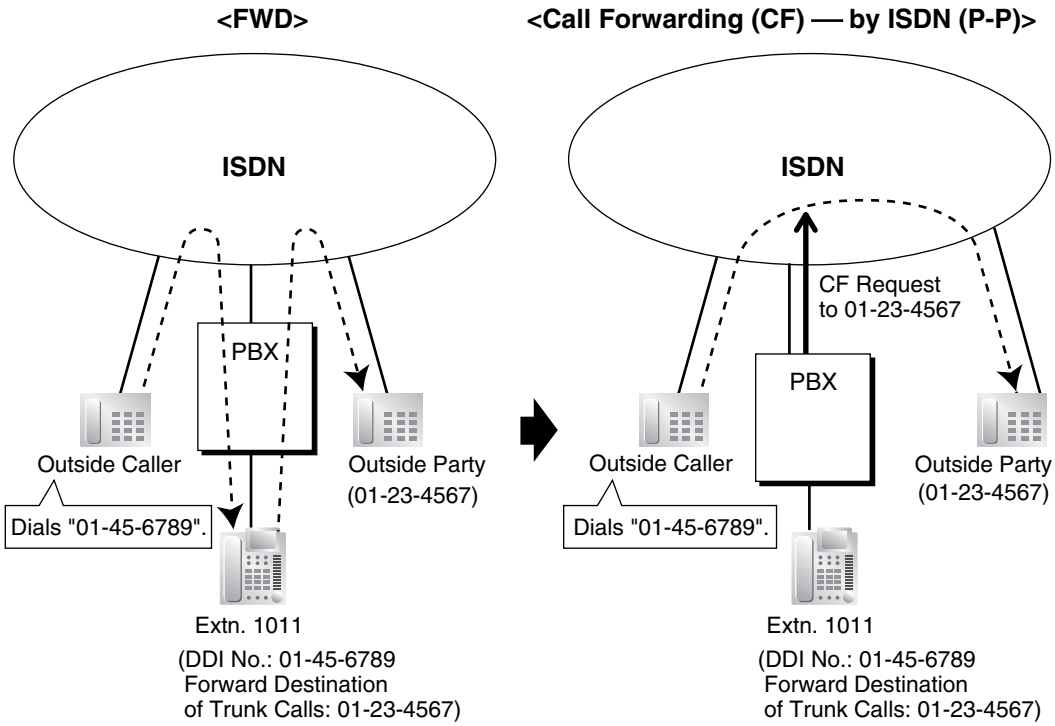
Description

An extension user can forward the incoming ISDN call to another outside party using the ISDN service of the telephone company, instead of the PBX feature, when the call is received through an ISDN line.
The network directly forwards the call to the destination which the extension user has set on the PBX as the forward destination of trunk calls; the network is instructed by the PBX. This feature is available only when the call is received through an ISDN port which supports this feature.

Call Forwarding—Unconditional (CFU), Call Forwarding—Busy (CFB), and Call Forwarding—No Reply (CFNR) are applied to this feature.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 207 Diversion supplementary service.

[Example]



Conditions

- The availability of this feature is dependent on the contract with the telephone company.
- This feature can be enabled or disabled on each ISDN port of the PBX.
- This feature is available when the same trunk group is used for the incoming call and the forwarded call.

Feature Guide References

1.3.1.2 Call Forwarding (FWD)

User Manual References

User Manual

1.5.1 Forwarding Calls

1.19.1.6 Call Hold (HOLD)—by ISDN

Description

An ISDN call can be put on hold using the ISDN service of the telephone company, instead of the PBX feature. This can be a part of a Call Transfer (CT)—by ISDN (→ 1.19.1.7 Call Transfer (CT)—by ISDN). This feature allows an ISDN call to be held, and a call to be made to another outside party using only one communication channel of ISDN. A PT user can easily use this feature by pressing the ISDN Hold button.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 141 Call Hold (HOLD) supplementary service.

Conditions

- **ISDN Hold Button**
  A flexible button can be customised as the ISDN Hold button.
- The availability of this feature is dependent on the contract with the telephone company.
- The TRS/Barring feature is applied when making a call after activating this feature. (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring))
- ARS cannot be applied to the call dialled after activating this feature. (→ 1.9.1 Automatic Route Selection (ARS))
- It is impossible to seize any other trunk during this feature.

Feature Guide References

1.18.2 Flexible Buttons

1.19.1.7 Call Transfer (CT)—by ISDN

Description

An ISDN call can be transferred to an outside party using the ISDN service of the telephone company, instead of the PBX feature, without occupying a second ISDN line.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 369 Explicit Call Transfer (ECT) supplementary service.

Conditions

- The availability of this feature is dependent on the contract with the telephone company.
- This feature can be enabled or disabled on an ISDN port basis.
- If an ISDN port is in P-P configuration, this feature can be used only when the network supports the "explicit linkage" option.
- Call Transfer with Announcement and Call Transfer without Announcement is possible. (→ 1.11.1 Call Transfer)
- The call charges after completing this feature will not be recorded by the PBX.

User Manual References

User Manual

1.4.1 Transferring a Call (Call Transfer)

1.19.1.8 Malicious Call Identification (MCID)

Description

An extension user can ask the telephone company to trace a malicious caller during a call or while hearing reorder tone after the caller hangs up. Information on the malicious call will be received later on.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 130 Malicious Call Identification (MCID) supplementary service.

User Manual References

User Manual

1.3.6 Identifying Malicious Calling Parties (Malicious Call Identification [MCID])

1.19.1.9 Completion of Calls to Busy Subscriber (CCBS)

Description

If the called party is busy and the call has been made using an ISDN line, an extension user can set to receive callback ringing when the called party becomes free. When the user answers the callback ringing, that party's number is automatically dialled.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 359 Completion of Calls to Busy Subscriber (CCBS) supplementary service.

Conditions

- This feature is available under the following conditions:
  - **a)** The caller's PBX is capable of using CCBS and the service is provided by the network.
  - **b)** The called party's PBX is capable of accepting CCBS.
- To receive and send CCBS, receiving and sending CCBS must be enabled individually on an ISDN port basis through system programming.
- An extension user can set only one CCBS. The last setting is effective.
- The CCBS setting is cancelled if there is no callback ringing within 60 minutes or callback ringing is not answered within 10 seconds.
- If an extension user makes a call using the CCBS feature, the preassigned CLIP number for the extension will be used even though the extension has set CLIR or the extension has used the CLIP for the incoming call distribution group by pressing the ICD Group button when the CCBS has set. (→ 1.19.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP))
- After using the CCBS feature, using Last Number Redial will not retrieve the number dialled by CCBS. (→ 1.6.1.4 Last Number Redial)
- An extension user that has set the CCBS feature cannot receive callback ringing while the extension is holding a call.

User Manual References

User Manual

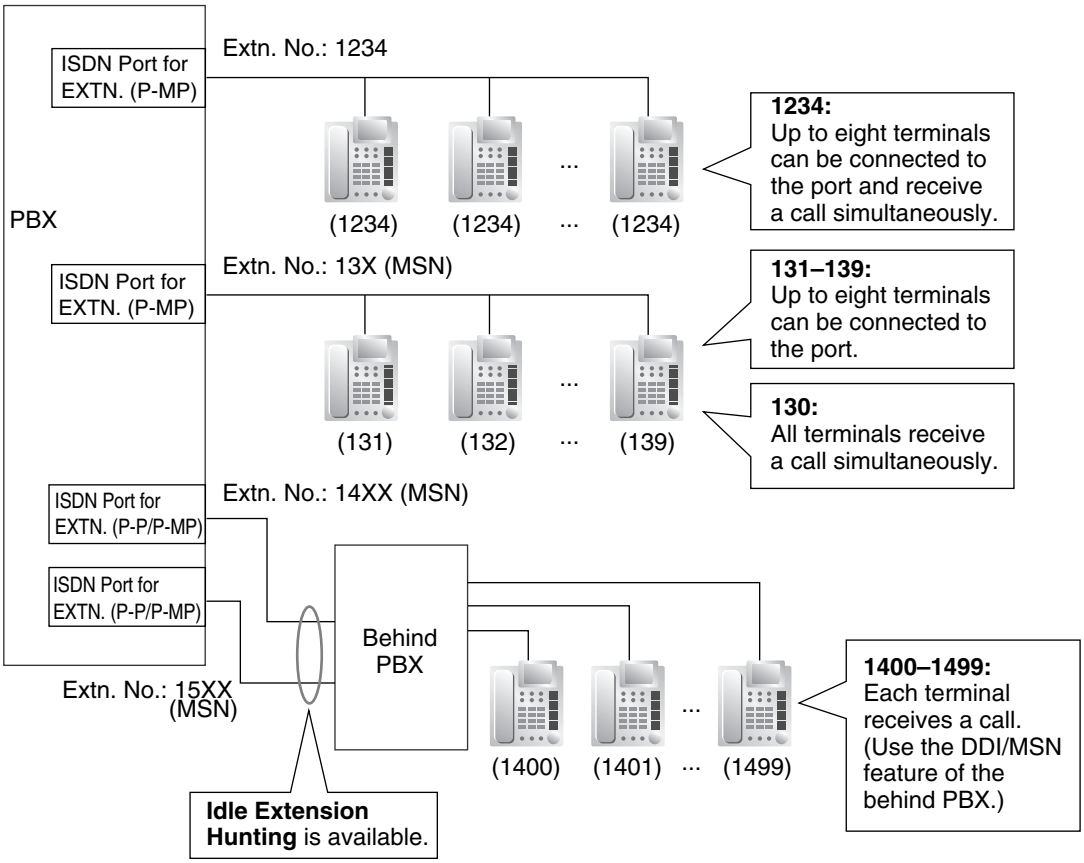
1.2.4 When the Dialled Line is Busy or There is No Answer

1.19.1.10 ISDN Extension

Description

An ISDN (BRI/PRI) port can be used for either trunk or extension connection. When extension connection is enabled, ISDN terminal devices (e.g., ISDN telephone, G4 fax machine, personal computer) or a behind PBX can be connected to the port.
If the ISDN port is in point-to-point (P-P) configuration, one terminal device can be connected to the port. If the ISDN port is in point-to-multipoint (P-MP) configuration, a maximum of eight terminal devices can be connected to the port. However, only a maximum of two devices can be used simultaneously.
Individual terminal devices on a behind PBX can be addressed with Multiple Subscriber Number (MSN). The MSN consists of the ISDN extension number and an additional digit, 0 through 9 or 00 through 99.

[Example]



Conditions

- **Subaddressing**
  Subaddressing is possible between ISDN terminal devices. The subaddress goes through the PBX to the ISDN terminal devices directly.
- Call Transfer (PBX feature) is available only for ISDN extensions in P-MP configuration. (→ 1.11.1 Call Transfer)

- ISDN extensions can belong to an incoming call distribution group (→ 1.2.2 Incoming Call Distribution Group Features) or idle extension hunting group (→ 1.2.1 Idle Extension Hunting). In this case, an MSN can be assigned.
- If the last digit of the MSN is "0", all terminal devices on the same ISDN port receive the call simultaneously, if the specification of each terminal device is available. To use the MSN whose last digit is "0" as an individual MSN, system programming is required for each ISDN port.

Feature Guide References

1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service

1.19.1.11 ISDN Service Access by Keypad Protocol

Description

ISDN provides some supplementary services by key protocol, and they may require a service access code to be dialled.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 122 Generic keypad protocol for the support of supplementary service (ISDN Service Access).

Conditions

- **ISDN Service Button**
  A flexible button can be customised as an ISDN Service button. A service access code can also be assigned on this button for a quick operation.
- This feature is not available to an SLT.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.2.5 Accessing the ISDN Service (ISDN Service Access)

1.20 E1 Line Service Features (KX-TDA100/KX-TDA200 only)

1.20.1 E1 Line Service (KX-TDA100/KX-TDA200 only)

Description

The E1 line carries thirty 64 kbps-voice channels at 2.048 Mbps transmission speed as a trunk or private line. Voice is digitised by Pulse Code Modulation (PCM).

1. **Channel Type**
   [DR2 (Digital System R2)/E & M-C (Continuous E & M)/E & M-P (Pulsed E & M)]
   The E1 card supports three different channel types to provide connection at minimum expense. One of the three channel types can be assigned to one of the 30 channels of the E1 card individually according to tariffs and customer needs.

2. **E1 Features**
   The following table shows the features available for each channel type:
   [Feature Table]

Channel Type	(1) TIE	(2) DID	(3) DIL	(4) ANI	(5) Call Charge Information
DR2		✔	✔	✔*1	✔
E & M-C	✔	✔	✔	✔*2	
E & M-P	✔	✔	✔	✔*2	

✔: Available
*1: Receiving Dial Mode should be MFC-R2.
*2: Available when this PBX is connected to the KX-TDA100, KX-TDA200, or KX-TD500.

[Explanation]

Number in the table	Feature	Description
(1)	**TIE Line Service**	Refer to the TIE Line Service feature. (→ 1.27.1 TIE Line Service)
(2)	**Direct Inward Dialling (DID)**	Refer to the DID feature. (→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI))
(3)	**Direct In Line (DIL)**	Refer to the DIL feature. (→ 1.1.1.2 Direct In Line (DIL))
(4)	**Automatic Number Identification (ANI)**	**Outgoing ANI:** Sends the caller's number to the E1 line. The sending method is the same as ISDN CLIP service. (→ 1.19.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)) **Incoming ANI:** Receives the caller's number from the E1 line. When the ANI number is received, it can be treated the same as a Caller ID number. (→ 1.16.1 Caller ID)
(5)	**Call Charge Information**	The call charge meter pulses can be received during a conversation. (→ 1.24.2 Charge Meter)

Conditions

- If an E1 line is used as a trunk, the channel type depends on the contract with the telephone company.
- If "MFC-R2" is selected as the Dial Mode, the PBX always sends a dial tone, instead of the telephone company, when making a trunk call using E1 line.

1.21 T1 Line Service Features (KX-TDA100/KX-TDA200 only)

1.21.1 T1 Line Service (KX-TDA100/KX-TDA200 only)

Description

The T1 line carries twenty-four 64 kbps-voice channels at 1.5 Mbps transmission speed as a trunk or private line. Voice is digitised by Pulse Code Modulation (PCM).

1. **Channel Type**

   [LCOT/GCOT/DID/TIE/OPX]
   The T1 card supports five different channel types to provide connection at minimum expense. One of the five channel types can be assigned to one of the 24 channels of the T1 card individually according to tariffs and customer needs.

2. **T1 Features**

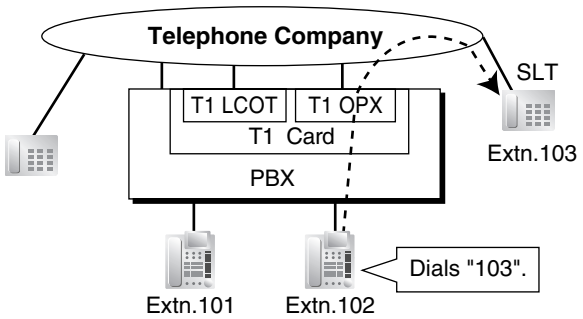
   The following table shows the features available for each channel type:
   [Feature Table]

Channel Type	(1) TIE	(2) DID	(3) DIL	(4) OPX	(5) ANI
LCOT			✔		
GCOT			✔		
DID		✔	✔		✔
TIE (E & M)	✔	✔	✔		✔
OPX				✔	

✔: Available

[Explanation]

Number in the Table	Feature	Description
(1)	**TIE Line Service**	Refer to the TIE Line Service feature. (→ 1.27.1 TIE Line Service)
(2)	**Direct Inward Dialling (DID)**	Refer to the DID feature. (→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI))
(3)	**Direct In Line (DIL)**	Refer to the DIL feature. (→ 1.1.1.2 Direct In Line (DIL))

Number in the Table	Feature	Description
(4)	**Off Premise Extension (OPX)**	An SLT which is installed off site can perform as if it were installed locally. The SLT makes and receives a call through the PBX. An extension number can be given to the OPX, and extension data such as COS is also applied. **[Example]** 
(5)	**Automatic Number Identification (ANI)**	Receives the caller's number from the T1 line. When the ANI number is received, it can be treated the same as a Caller ID number. (→ 1.16.1 Caller ID)

Conditions

- If a T1 line is used as a trunk, the channel type depends on the contract with the telephone company.

1.22 Voice Mail Features

1.22.1 Voice Mail (VM) Group

Description

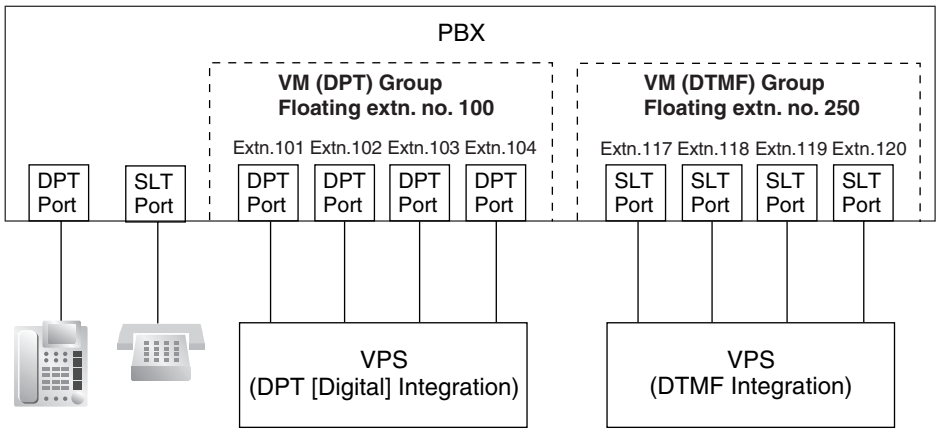
A VPS can be connected to extension ports on the PBX. The extension ports make a group, called a VM group. This group has a floating extension number. The VM group can be the destination for redirected calls and incoming calls. When receiving a call, the VPS can greet the caller offering them the option to leave a message or dial a number to reach the desired party. The VPS can record the message for each extension and leave notification on the corresponding extension, if the called extension is not able to answer calls.

1. VM Group Type

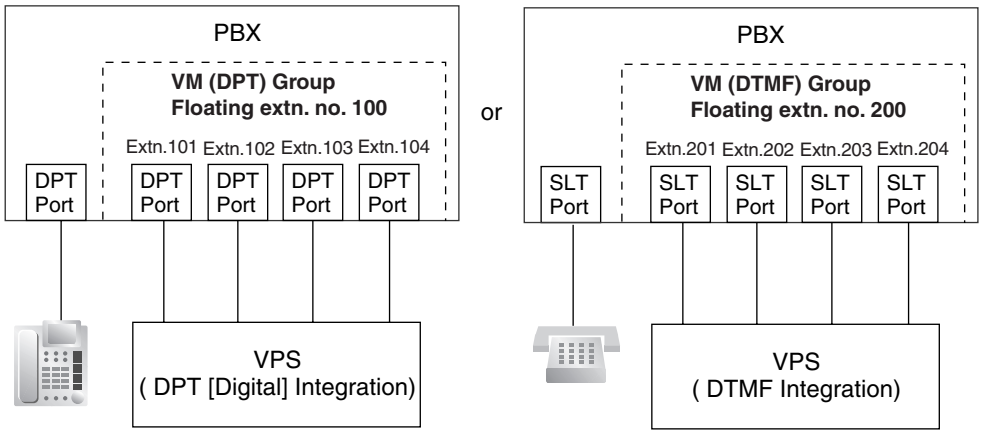
Type	Description
VM (DTMF) Group	A group of SLT ports which use the Voice Mail DTMF Integration features. A maximum of two VM (DTMF) groups can be assigned.
VM (DPT) Group	A group of DPT ports which use the Voice Mail DPT (Digital) Integration features. With the KX-TDA100/KX-TDA200: • A maximum of two VM (DPT) groups can be assigned. • One Panasonic KX-TVS/KX-TVP system can be in only one group. • A maximum of 12 ports (24 channels) of the VPS can form each group. With the KX-TDA30: • One VM (DPT) group can be assigned. • A maximum of 4 ports (8 channels) of the VPS can form a group.

[Example]

<KX-TDA100/KX-TDA200>



<KX-TDA30>



VM (DTMF) Group Assignment:

The VPS is connected to the SLT ports of the PBX. These SLT ports, as well as VM (DTMF) Group settings, must be configured to allow DTMF Integration, as shown in the following [Programming Example of Extension Port] and [Programming Example of VM (DTMF) Group].

VM (DPT) Group Assignment:

The VPS is connected to the DPT ports of the PBX. These DPT ports, as well as VM (DPT) group settings, must be configured to allow DPT (Digital) Integration, as shown in the following [Programming Example of Extension Port] and [Programming Example of VM (DPT) Group].

[Programming Example of Extension Port]

Slot	Port	Port Type	Extn. No.	DPT Property			
				Type*	Unit No. of VPS	Port No. of VPS	
3	1	DPT	101	VM (DPT)	1	1	VM (DPT) Group 1
3	2	DPT	102	VM (DPT)	1	2	
:	:	:	:	:	:	:	
4	1	S-Hybrid	201	VM (DPT)	2	1	VM (DPT) Group 2
4	2	S-Hybrid	202	VM (DPT)	2	2	
:	:	:	:	:	:	:	
5	1	SLT	301	Not assignable	Not assignable	Not assignable	VM (DTMF) Group 1
5	2	SLT	302	Not assignable	Not assignable	Not assignable	
:	:	:	:	:	:	:	
6	1	SLT	401	Not assignable	Not assignable	Not assignable	VM (DTMF) Group 2
6	2	SLT	402	Not assignable	Not assignable	Not assignable	

*: → Terminal Device Assignment [601]

[Programming Example of VM (DTMF) Group]

VM (DTMF) Group No.	Floating Extension No.	Group Name	Service Mode	Extension No. of Port Connected to VPS Port		
				1	2	…
1	300	Company C	AA	301	302	…
2	400	Company D	VM	401	402	…

[Programming Example of VM (DPT) Group]

VM (DPT) Group No.	Floating Extension No.*	Group Name
1	100	Company A
2	200	Company B

*: → VM Group Floating Extension Number [660]

2. Incoming Calls to VM Group

When incoming calls are received at the floating extension number of the VM group, calls will hunt starting at the lowest VM port number. In this case, the FWD and DND settings (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND)) for each extension port are disregarded.

It is programmable whether the calls queue when all extension ports in the group are busy through system programming. If the queuing is disabled through system programming, the call will be redirected to the preprogrammed destination by the Intercept Routing feature.

Conditions

[General]

- It is possible to call an extension (extension port) in a VM group directly. If the calls are routed directly to the extension in the group, it is possible to enable some features (e.g., FWD, Idle Extension Hunting) on the extension in the group.
  The Voice Mail DTMF/DPT (Digital) Integration (e.g., command transmit) is also available on the extension.

- **One-touch Voice Mail Feature Access**
  It is possible to assign a One-touch Dialling button for direct access to a Voice Mail feature. (→ 1.6.1.2 One-touch Dialling) For example, to access a mailbox (mailbox number 123) of the VPS (extension number 165) directly, assign "**165**#6**123**" to a One-touch Dialling button. When pressing this button, the outgoing message (OGM) of the mailbox will be heard.

[KX-TDA100/KX-TDA200]

- All ports in a VM (DPT) group must be connected to a single DHLC or DLC card.

[KX-TDA30]

- All ports in a VM (DPT) group must be connected to either the same DLC card, or to the preinstalled super hybrid ports.

Feature Guide References

1.22.2 Voice Mail DTMF Integration

1.22.3 Voice Mail DPT (Digital) Integration

1.22.2 Voice Mail DTMF Integration

Description

The PBX and the VPS connected to the PBX can transmit commands using DTMF signals to each other. The PBX sends preprogrammed commands using DTMF to the VPS automatically to change the answering service between Voice Mail (VM) service mode and Automated Attendant (AA) service mode or to inform the extension status (e.g., busy). The VPS sends the commands to the PBX like an SLT.
The following answering services and features are available:

1. **Voice Mail (VM) Service Mode**
   When a caller reaches the VPS, the VPS greets and guides the caller to leave a voice message for a specified mailbox.
   [Example]
   The VPS sends the message to the caller, "Thank you for calling Panasonic. Please enter the mailbox number of the person you wish to leave your message for."
   ↓
   The caller dials the mailbox number. Then, the dialled number is sent to the VPS via the PBX.
   ↓
   The VPS sends the personal greeting to the caller, "You have reached Mike's voice mail. I am sorry I cannot take your call right now. Please leave a message and I will call you back."
   ↓
   The caller leaves a message.

   If the call reroutes to the floating extension number of the VM (DTMF) Group or the extension in the VM (DTMF) group by such as the FWD feature, when the VPS answers the call, the PBX will dial the mailbox number of the corresponding group or extension and any other digits required to the VPS automatically using the caller-dialled number (Follow on ID). In this case, the caller can reach a mailbox without knowing the mailbox number.
   [Available Features for Follow on ID]
   - **a)** FWD to a Mailbox
   - **b)** Intercept Routing to a Mailbox
   - **c)** Call Transfer to a Mailbox
   - **d)** Listening to a Message in a Mailbox

2. **Automated Attendant (AA) Service Mode**
   The VPS greets and guides the caller to the desired extension directly without operator assistance.

3. **VM → AA Service, AA → VM Service**
   It is possible to switch the service mode assigned on the VPS port, from the VM service to AA service or vice versa.

System Explanation

1. **Service Mode Assignment**
   Assign the service mode, VM service or AA service, to the VM (DTMF) group to

correspond with the assignment of the VPS.

2. DTMF Command Assignment

Assign the DTMF command to suit the VPS settings.

	Command (Default)
Switching to VM	#6
Switching to AA	#8
Recording message	H
Listening message	✳H

Note: H = Mailbox Number

3. VM Service

a) FWD to a Mailbox of the VPS

The PBX sends a mailbox number of the corresponding extension to the VPS when a call is forwarded from an extension to the VPS. Therefore the caller can leave a message for the called extension without knowing the mailbox number.

[FWD to the VPS Sequence Selection]

If an extension user sets FWD to the VPS, any incoming call will be forwarded to the VPS.

It is also possible to send the AA command, even in the VM service mode, when calls are forwarded so that the caller can be directed to an extension rather than a mailbox. This can be performed by selecting "AA" through system programming.

Parameter	Transmitted Command	
	In AA Service Mode	**In VM Service Mode**
Answer by Mailbox (Default)	Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])	Recording message command (H [H = Mailbox No.])
AA	Switching to AA command (#8)	Switching to AA command (#8)
None	(DTMF commands are not sent. Work with default of the VPS.)	

b) Intercept Routing to a Mailbox of the VPS

The PBX sends a mailbox number of the corresponding extension to the VPS when a trunk call is intercepted from an extension to the VPS. Therefore the caller can leave a message for the called extension without knowing the mailbox number.

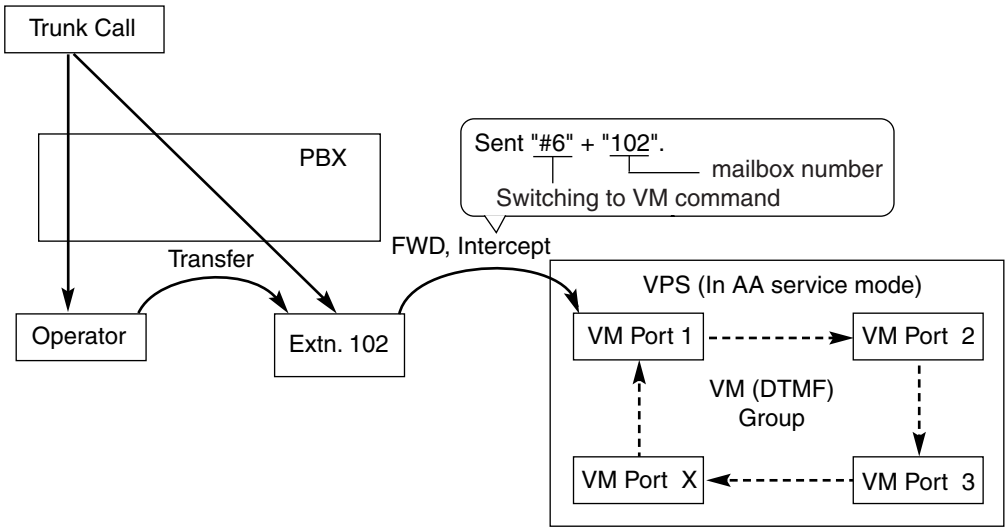
[Intercept Routing to the VPS Sequence Selection]

If an extension user sets Intercept Routing to the VPS, the intercepted trunk calls will be redirected to the VPS.

It is also possible to send the AA command, even in the VM service mode, when calls are intercepted so that the caller can be directed to an extension rather than a mailbox. This can be performed by selecting "AA" through system programming.

Parameter	Transmitted Command	
	In AA Service Mode	In VM Service Mode
Answer by Mailbox	Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])	Recording message command (H [H = Mailbox No.])
AA	Switching to AA command (#8)	Switching to AA command (#8)
None (Default)	(DTMF commands are not sent. Work with default of the VPS.)	

[Example of a) & b)]



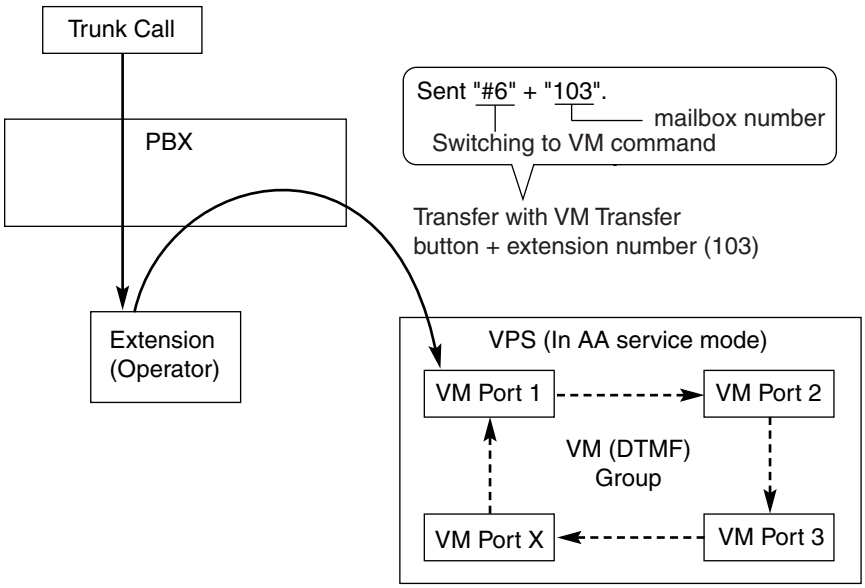
c) Transferring to a Mailbox of the VPS

An extension user can transfer a call to a mailbox of the VPS so that the caller can leave a message in the mailbox of the desired extension. The extension user should use the Voice Mail (VM) Transfer button, when transferring a call to the mailbox of the VPS. When the extension user presses this button and enters the desired extension number, the PBX will transfer the call to the VM group and dial the mailbox number of the desired extension with the required command (after the VPS answers the call). Therefore the caller can leave a message without knowing the mailbox number.

[Performance of Pressing the VM Transfer Button and Entering an Extension Number]

Transmitted Command	
In AA Service Mode	**In VM Service Mode**
Switching to VM command + Recording message command (#6 + H [H = Mailbox No.])	Recording message command (H [H = Mailbox No.])

[Example]



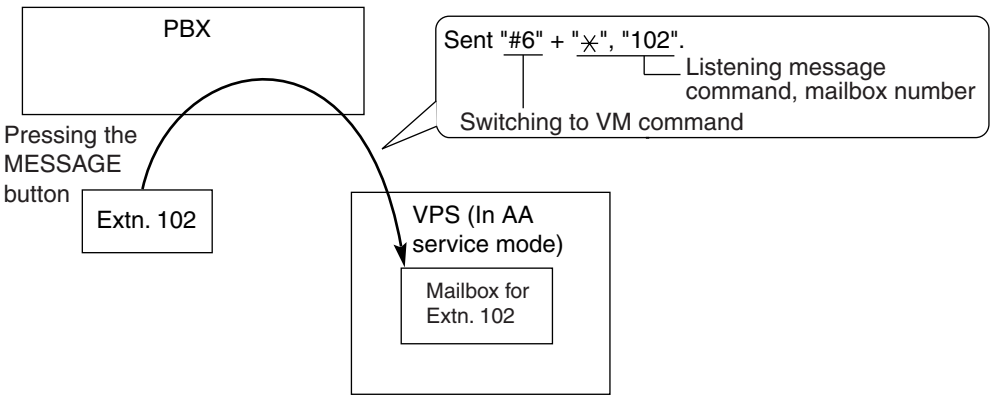
d) Listening to a Recorded Message

If the VPS receives a message, the VPS will set the Message Waiting feature (e.g., turning on the Message button light) on the corresponding telephone as notification. (→ 1.17.1 Message Waiting) Thereby, the VPS notifies the extension user that there is a message waiting in his mailbox. When the Message button light turns on, pressing the button allows the extension user to play back the messages stored in his mailbox without dialling such as a mailbox number. It is programmable whether the PBX or the VPS cancels the Message Waiting feature (e.g., turning off the Message button light). When the PBX is selected, the Message Waiting feature is cancelled after the extension user has accessed the VPS. When the VPS is selected, the Message Waiting feature is cancelled after the extension user has listened to messages stored in his mailbox.

[Performance of Pressing the MESSAGE Button]

Transmitted Command	
In AA Service Mode	**In VM Service Mode**
Switching to VM command + Listening message command (#6 + ⁎H [H = Mailbox No.])	Listening message command (⁎H [H = Mailbox No.])

[Example]



4. AA Service

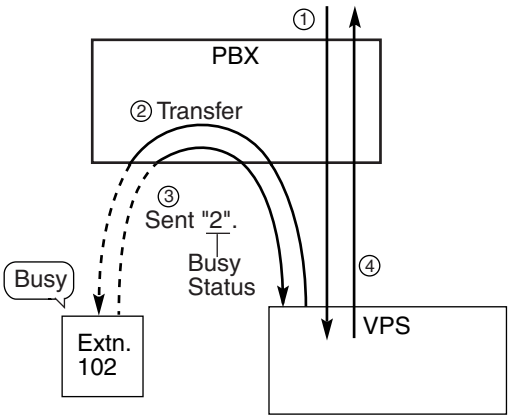
If the VPS transfers the call using the AA service, the PBX will inform the VPS of the status of the called destination with the preprogrammed DTMF status signal so that the VPS can confirm the status of the extension without listening to the system tones (e.g., ringback tone).
Assign the DTMF status signal to suit the VPS settings.

[The DTMF Status Signal and Condition]

Status	Condition	Default Command
RBT (ringback tone)	The PBX is ringing the corresponding extension.	1
BT (busy tone)	The called extension is busy.	2
ROT (reorder tone)	The dialled number is invalid.	3
DND (DND tone)	The called extension has set DND. (→ 1.3.1.3 Do Not Disturb (DND))	4
Answer	The called extension has answered the call.	5
FWD VM RBT (FWD to Voice Mail ringback tone)	The called extension has set FWD to VPS and the PBX is calling another port of the VPS.	6
FWD VM BT (FWD to Voice Mail busy tone)	The called extension has set FWD to VPS and all ports of the VPS are busy.	7

Status	Condition	Default Command
FWD EXT RBT (FWD to extension ringback tone)	The PBX is calling an extension other than the one dialled. FWD or Idle Extension Hunting (→ 1.2.1 Idle Extension Hunting) may be assigned by the called extension.	8
Confirm (confirmation tone)	The PBX receives confirmation that the feature has been successfully set or cancelled (e.g., Message Waiting) on the extension.	9
Disconnect	The caller has hung up.	#9

[Example]



① An incoming call reaches the VPS. The VPS greets the caller: "Thank you for calling Panasonic. If you know the extension number of the person you wish… ".

② The caller dials the extension number (extn. 102). The VPS will transfer the call to the extension via the PBX.

③ If the extension is not available, the PBX sends DTMF status signal of the extension (busy status) to the VPS.

④ The VPS receives the DTMF status signal and send the appropriate message to the caller: "Sorry. The extension is busy. Would you like to leave a message… ".

Conditions

- **Voice Mail (VM) Transfer Button**
  A flexible button can be customised as the Voice Mail (VM) Transfer button with the floating extension number of the VM group as the parameter.
- It is possible to assign the time period between when the VPS answers the call and the PBX sends the Follow on ID to the VPS.
- It is possible to assign the time period between when the VPS transfers the call using the AA service to the PBX and the PBX sends the DTMF status signal to the VPS.
- It is possible to select whether the mailbox number is the same as the extension number, or the mailbox number is programmable for each extension number and incoming call distribution group (Mailbox Access ID).
- The Inter-digit time for the DTMF command and for DTMF status signal is programmable.
- Data Line Security is set automatically on the extensions in the VM (DTMF) group to achieve proper recording. (→ 1.10.5 Data Line Security)

Feature Guide References

1.1.1.6 Intercept Routing

1.3.1.2 Call Forwarding (FWD)
1.11.1 Call Transfer
1.18.2 Flexible Buttons
1.22.1 Voice Mail (VM) Group

User Manual References

User Manual

1.9.3 If a Voice Processing System is Connected

1.22.3 Voice Mail DPT (Digital) Integration

Description

A Panasonic VPS that supports DPT (Digital) Integration (e.g., the KX-TVP200) can be connected to this PBX in a tightly integrated fashion.
DPT (Digital) Integration features can be used when the VPS is connected through DPT ports of the PBX.

Feature Explanation

1. **Automatic Configuration—Quick Setup**

   The PBX informs the VPS of its extension numbers and the floating extension numbers of the incoming call distribution groups so that the VPS can create mailboxes with this data automatically.

2. **FWD to a Mailbox of the VPS**

   If an extension user sets FWD to the VPS, incoming calls are forwarded to the VPS. (→ 1.3.1.2 Call Forwarding (FWD)) The PBX sends a mailbox number of the forwarding extension to the VPS. Therefore the caller can leave a message in the mailbox of the extension without knowing the mailbox number.

3. **Intercept Routing to a Mailbox of the VPS**

   If an extension user sets Intercept Routing to the VPS, the intercepted trunk calls will be redirected to the VPS. (→ 1.1.1.6 Intercept Routing) If the "`Intercept to Mailbox setting`" is enabled through system programming, the PBX sends the mailbox number of the intercepted extension to the VPS. Therefore the caller can leave a message in the mailbox of the extension without knowing the mailbox number.

4. **Transferring to a Mailbox of the VPS**

   An extension user can transfer a call to a mailbox of the VPS so that caller can leave a message in the mailbox of the desired extension. (→ 1.11.1 Call Transfer) The extension user should use the Voice Mail (VM) Transfer button, when transferring a call to the mailbox of the VPS. When the extension user presses this button and enters the desired extension number, the PBX will transfer the call to the VM group and dial the mailbox number of the desired extension number with the required command (after the VPS answers the call). Therefore the caller can leave a message without knowing the mailbox number.

5. **Transfer Recall to a Mailbox of the VPS**

   If a call is transferred to an extension via the Automated Attendant (AA) service of the VPS and the call is not answered within a preprogrammed Transfer Recall time (→ Transfer Recall Time [201]), the PBX sends the mailbox number of the transfer destination extension to the VPS. Therefore the caller can leave a message in the mailbox of the extension without knowing the mailbox number. The "`Transfer Recall to Mailbox`" setting should be enabled through system programming to use this feature.

6. **Listening to a Recorded Message (Direct Mailbox Access)**

   If the VPS receives a message, the VPS will set the Message Waiting feature (e.g., turning on the Message button light) on the corresponding telephone as notification. (→ 1.17.1 Message Waiting) Thereby, the VPS notifies the extension user that there is a message waiting in his mailbox. When the Message button light turns on, pressing the button allows the extension user to play back the messages stored in his mailbox without dialling such as a mailbox.
   When the extension user dials an extension number of the VM (DPT) extension port or the floating extension number of the VM (DPT) group from his extension, he can listen to the

messages stored in his mailbox without dialling his mailbox number (Direct Mailbox Access). It is possible to disable this feature by COS programming on the VPS.

7. VPS Trunk Service & Automatic Time Mode Notification for Incoming Call*

Multiple tenants can share a single VPS; each tenant does not require a dedicated VPS port. If the destination of the incoming trunk call is a VM (DPT) group, the PBX sends the VPS trunk group number and time mode (day/lunch/break/night) of the tenant (→ 2.2.4 Time Service) assigned for the call to the VPS. Therefore the VPS can send the assigned message (company greeting) to the caller. Corresponding VPS trunk group number and tenant number are determined by the setting of the incoming trunk call as follows:

a) **DIL/TIE:** the setting of each trunk port (→ 1.1.1.2 Direct In Line (DIL), 1.27.1 TIE Line Service)

b) **DID/DDI:** the setting of each location number for DID/DDI (→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI))

c) **MSN:** the setting of each MSN (→ 1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service)

[Example]
For DID Calls:
Each location number can have its VPS trunk group number (message number) and tenant number.

[DID Programming Example]

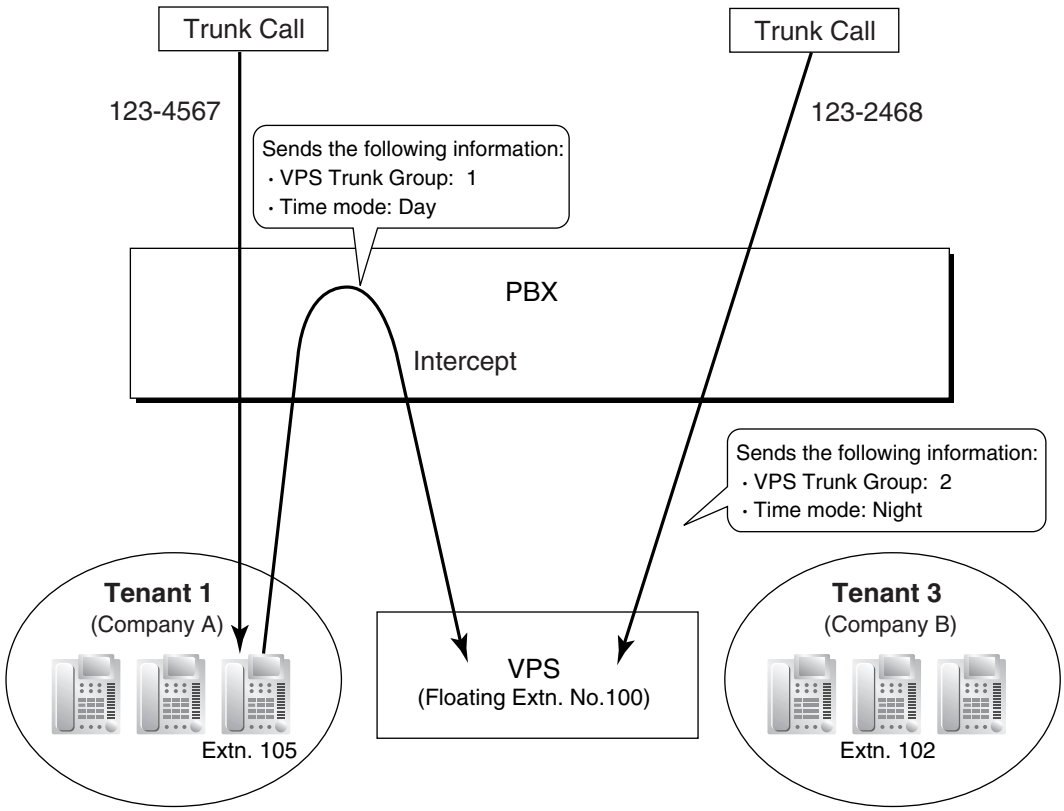
Location No.	DID No.	DID Destination				DID Name	VPS Trunk Group No.	Tenant No.
		Day	Lunch	Break	Night			
0001	123-4567	105	100	105	100	John White	1	1
0002	123-2468	102	100	102	100	Tom Smith	2	3
:	:	:	:	:	:	:	:	:

[VPS Programming—Programming Example of Trunk Group Assignment]

Trunk Group No.		Company Greeting No.	Incoming Call Service	. .
1	Day	1	Custom Service 11	. .
	Lunch	2	Custom Service 29	. .
	Break	3	Custom Service 31	. .
	Night	4	Custom Service 12	. .
2	Day	5	Custom Service 21	. .
	Lunch	6	Custom Service 15	. .
	Break	7	Custom Service 42	. .
	Night	8	Custom Service 30	. .
:	:	:	:	. .

Explanation:
A DID call reaches a VM (DPT) group directly or by the Intercept Routing feature. According to the [DID Programming Example] and [VPS Programming—Programming Example of Trunk Group Assignment], a caller will hear a corresponding company greeting of the VPS.

Time mode (day/lunch/break/night) of the preprogrammed tenant is applied to the DID destination and company greeting number.



8. Caller's Identification Notification to the VPS

When receiving a trunk call, the PBX sends the caller's identification number/name to the VPS.

9. Status Notification to the VPS

After the call is redirected by the VPS, the PBX sends the status of the redirected extension (e.g., busy) to the VPS.

10. Paging by the VPS

The VPS can perform the Paging feature using the recorded message. (→ 1.14.1 Paging)

11. Live Call Screening (LCS)

PT and PS users can monitor his own mailbox while a caller is leaving a message and, if desired, answer the call by pressing the LCS button. When the caller is leaving a message in the mailbox, monitoring can be carried out in two ways: each PT user can choose which through personal programming (Live Call Screening Mode Set). PS users cannot choose the way: only Private mode is available for them.
Hands-free mode: The user can monitor the call automatically through the built-in speaker.
Private mode: The user will hear an alarm tone. To monitor the call, the user goes off-hook with the handset, MONITOR button, or SP-PHONE button. However, PS users cannot monitor the call with the speakerphone.

12. Two-way Recording into the VPS

A PT user can record a conversation into his own mailbox or another mailbox, while talking on the phone.

Use the Two-way Record button to record into his own mailbox. Use the Two-way Transfer button to record into someone else's mailbox.

Note

When you record Two-way telephone conversations, you should inform the other party that the conversation is being recorded.

13. **VPS Data Control by the PBX***

    The data of date and time is controlled by the PBX.

14. **Remote PBX Data Control by the VPS***

    The FWD of an extension can be controlled by the VPS.

*: This feature may not be supported depending on the software version of the VPS.

Conditions

[General]

- **Voice Mail (VM) Transfer Button**
  A flexible button can be customised as the Voice Mail (VM) Transfer button with the floating extension number of the VM group as the parameter.

[Live Call Screening (LCS)]

- If an SLT is connected in parallel to a PT, and if LCS is activated for the PT in Private mode, both the PT and SLT can be used to monitor calls while in idle status. The SLT will ring to indicate a message is being recorded. The call can be monitored with the SLT by going off-hook. To intercept the call, press Flash/Recall button or flash the hookswitch. (→ 1.10.9 Parallelled Telephone)
- **LCS Button**
  A flexible button can be customised as the LCS button.
- **Extension Personal Identification Number (PIN)**
  To prevent unauthorised monitoring, it is recommended the LCS user assign an extension PIN. This PIN will be required when setting LCS. (→ 1.25.1 Extension Personal Identification Number (PIN)) If the user forgets the PIN, it can be cleared by an extension assigned as the manager.
- Each extension can be programmed to either end recording or continue recording the conversation after the call is intercepted, through personal programming (LCS Mode Set [After Answering]).
- To use the LCS feature on a PS in Wireless XDP Parallel Mode, LCS can only be turned on or off from the wired telephone. In Wireless XDP Parallel Mode, setting LCS on/off from the PS has no effect. (→ 1.23.5 Wireless XDP Parallel Mode)

[Two-way Recording into the VPS]

- **Two-way Record/Two-way Transfer Button**
  A flexible button can be customised as the Two-way Record or the Two-way Transfer button. An extension number can be assigned to the Two-way Transfer button so that it can be used as a one-touch record button for the mailbox of the specified extension. (**One-touch Two-way Transfer Button**).
- When all of the VPS ports are busy:

  a) Pressing the Two-way Record button sends an alarm tone

b) Pressing the Two-way Transfer button followed by an extension number sends an alarm tone.

Feature Guide References

1.18.2 Flexible Buttons
1.22.1 Voice Mail (VM) Group

User Manual References

User Manual

1.9.3 If a Voice Processing System is Connected
3.1.2 Settings on the Programming Mode
3.2.2 Manager Programming

1.23 Portable Station (PS) Features

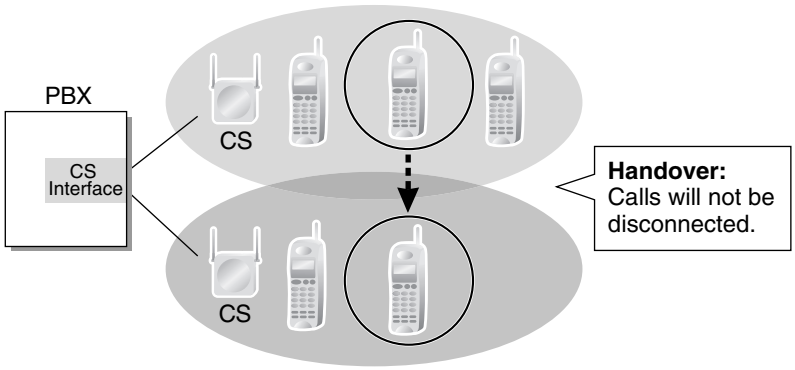
1.23.1 Portable Station (PS) Connection

Description

This PBX supports the connection of a PS (e.g., KX-TD7590, KX-TD7690). It is possible to use the PBX features using the PS like a PT.

Conditions

- The PS registration is required through the system programming (→ PS Registration [690]). To avoid unexpected registration to another PBX, the Personal Identification Number (PIN) for the PBX is necessary to register a PS (→ Personal Identification Number (PIN) for PS Registration [692]). The registration can be cancelled (→ PS Termination [691]).
- **Handover**
  Even if a PS user moves during a conversation, the PS will automatically switch between cells without disconnecting the call (**Handover**).
  Handover is available in any of the following cases:
  a) During a conversation with an extension or outside party.
  b) While a call is ringing at the PS.
  c) While the PS is in idle status.



On the contrary, Handover is not available in one of the following cases:
a) When the new (Handover) CS is busy.
b) When there is no CS within range.
c) During a Conference call (→ 1.13.1 Conference Features).
d) While the Live Call Screening (LCS) or Two-way Record is activated (→ 1.22.3 Voice Mail DPT (Digital) Integration).
e) While the PS user is paging other extensions (→ 1.14.1 Paging).
f) During a doorphone call (→ 1.15.1 Doorphone Call).
g) While the PS user is dialling digits to make a trunk call.
h) During a conversation with an extension using PC Console or PC Phone, to which the conversation is being recorded.

- When a caller has dialled the extension number of a PS but the CS is busy, the caller hears busy tone.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.8 Connection of DECT Portable Stations — KX-TDA100/KX-TDA200

2.9 Connection of 2.4 GHz Portable Stations — KX-TDA100/KX-TDA200

<KX-TDA30>

6.7 Connection of DECT Portable Stations — KX-TDA30

6.8 Connection of 2.4 GHz Portable Stations — KX-TDA30

1.23.2 PS Ring Group

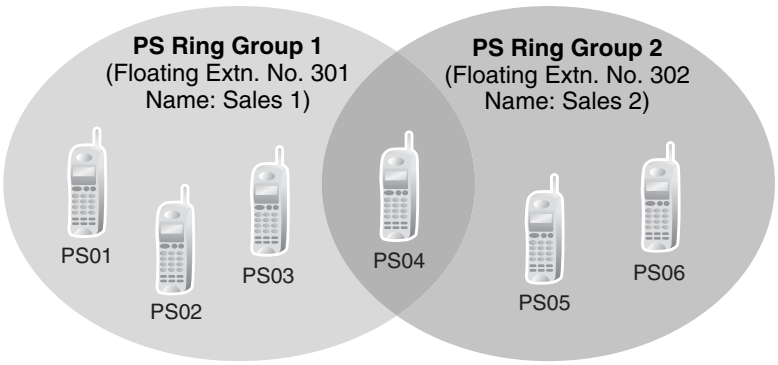
Description

PS ring group is a group of PS extensions which receives incoming calls. Each group has a floating extension number and name. One PS can belong to multiple groups.

[Programming Example]

	PS Ring Group 01	PS Ring Group 02	PS Ring Group 03	. .
Floating Extn. No.	301	302	303	. .
Group Name	Sales 1	Sales 2	Sales 3	. .
Incoming Trunk Call Information Display	Called Party's Name/ Number	Caller's Name/ Number	Caller's Name/ Number	. .
PS01	✔			. .
PS02	✔			. .
PS03	✔			. .
PS04	✔	✔		. .
PS05		✔		. .
PS06		✔		. .
PS07			✔	. .
:	:	:	:	:

✔ : Constituent



Conditions

- **PS Ring Group**
  A maximum of 32 groups can be created.
- **Available PS**
  The KX-TD7580, KX-TD7590, KX-TCA155, KX-TCA255, KX-TD7680, and KX-TD7690 can join PS ring groups.
- Incoming trunk call information is shown on a PS display when a trunk call arrives at a PS ring group which the PS joins. The display information can be selected on a PS ring group

basis through system programming: Called Party's Name/Number or Caller's Name/Number.

- **Calling Multiple PSs Simultaneously**

  There are two methods to call multiple PSs simultaneously using the floating extension number assigned to the following groups:

Method	Assignment	Merit	Demerit	Details in
Incoming Call Distribution Group	Assign all desired PSs to one incoming call distribution group (→ Incoming Call Distribution Group Member [620]), and set the group call distribution method for the group to "Ring".	All PS users in the group can use the Log-in/Log-out feature, Wrap-up feature, and ICD Group button for the group.	The CS may often be busy as each PS in the group uses one channel when a call arrives at the group.	• 1.2.2 Incoming Call Distribution Group Features
PS Ring Group	Assign all desired PSs to one PS ring group.	Only one channel is used when a call arrives at the group.	PS users in the group cannot use the Log-in/Log-out and Wrap-up features.	–

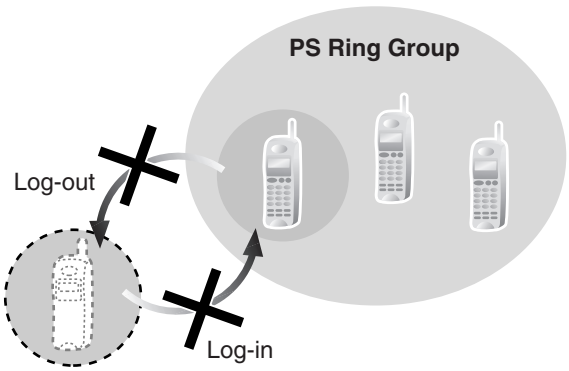
- When a PS joins a PS ring group, the following personal settings are disregarded:

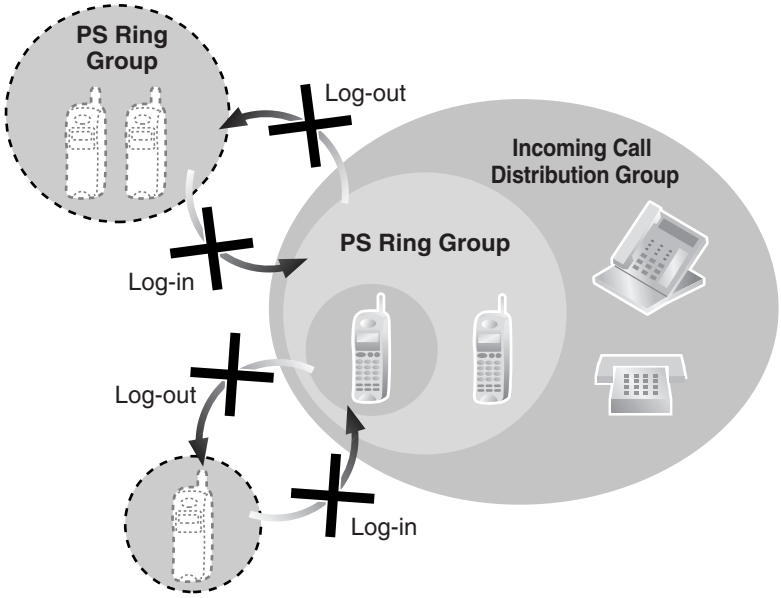
  a) When the PS ring group is called:

  – Delayed Ringing
  – Display information when the incoming calls arrive;
    The settings (e.g., display priority) are disregarded.
  – The setting which is assigned on the PS (e.g., FWD)
  – The status of the PS (e.g., busy)

  b) Log-in/log-out setting (from the PS ring group/from the incoming call distribution group which the PS ring group belongs to). (→ 1.2.2.6 Log-in/Log-out)

  Note
  Log-in/log-out setting of the PS ring group from the incoming call distribution group is also disregarded.





- When the PS ring group is called using the floating extension number, the group becomes busy to other callers using the floating extension number. However, the individual group members may be called directly using their extension number.
- If a PS in a PS ring group has set the DND feature for trunk calls, the PS will not ring when an intercom call or a trunk call arrives at the PS ring group. (→ 1.3.1.3 Do Not Disturb (DND))
- For calls directed to PS ring groups, the PBX will handle at most two calls simultaneously. The third call cannot arrive at a PS ring group until one of the first two calls is answered or a caller hangs up.

1.23.3 PS Directory

Description

A PS user can store numbers and/or names in the directory. A stored number is dialled by selecting a name or number in the directory.
Depending on the PS model, the PS user can use the following directories for easy operation:

Type	Description
PS Dialling Directory	Makes a call by selecting from a private directory of names and telephone numbers.
System Speed Dialling Directory	Makes a call by selecting from a common directory of names (→ System Speed Dialling Name [002]) and numbers (→ System Speed Dialling Number [001]).
PBX Extension Dialling Directory	Makes a call by selecting from a common directory of extension names (→ Extension Name [004]).
Shortcut Directory	Accesses a feature by selecting from a private directory of feature names and numbers.
Quick Dialling	Makes a call or accesses a feature easily by selecting from a private directory of names and numbers.

1.23.4 PS Feature Buttons

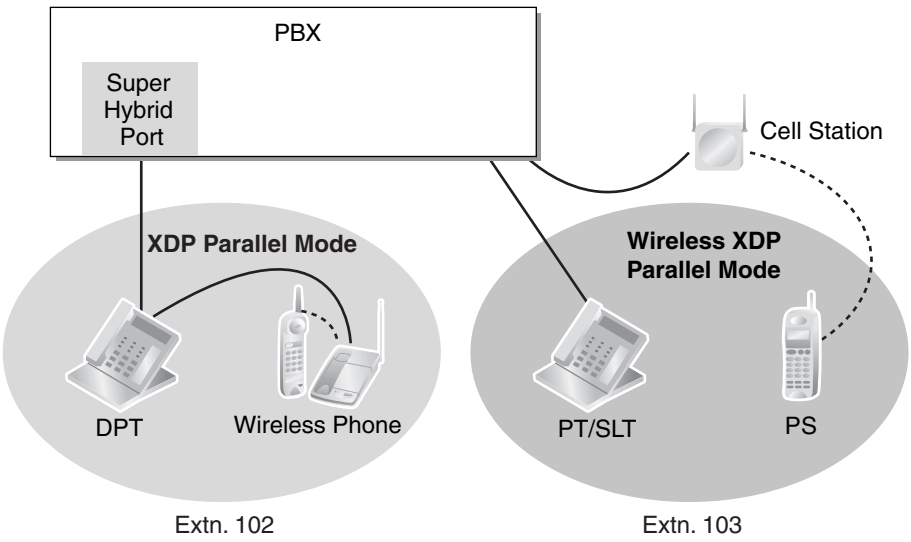
Description

A PS user can use PBX feature using a combination of buttons (button + a specified number, ✳, or #) and/or display operation. The flexible buttons and the display are customised through PS programming. The button assignment is the same for PT. (→ 1.18.2 Flexible Buttons). Some special feature buttons (e.g., WAVESEARCH button) may be customised depending on the PS type.

1.23.5 Wireless XDP Parallel Mode

Description

A PS can be used in parallel with a wired telephone (PT/SLT). In this case, the wired telephone is the main telephone and the PS is the sub telephone. When Wireless XDP Parallel Mode is enabled, the two telephones share one extension number (main telephone's extension number) like XDP Parallel mode for a DPT and an SLT.



Conditions

- If one of the telephones goes off-hook while the other telephone is on a call, the call is switched to the telephone going off-hook. However, the call is not switched in one of the following conditions:
  - **a)** During a Conference call (→ 1.13.1 Conference Features).
  - **b)** While Live Call Screening (LCS) or Two-way Record is activated (→ 1.22.3 Voice Mail DPT (Digital) Integration).
  - **c)** While receiving OHCA (→ 1.7.4.3 Off-hook Call Announcement (OHCA)).
  - **d)** While being monitored by another extension (→ 1.7.3 Call Monitor).
  - **e)** During Consultation Hold.

  (**Consultation Hold:** Allows an extension user to place a call on hold temporarily to perform Call Transfer, Conference, or Call Splitting.)
- Wireless XDP Parallel Mode can only be set from a PS. The wired telephone can accept or deny this feature through COS programming (→ Wireless XDP Parallel Mode for Paired Telephone [515]). Once this feature is set, the setting at the wired telephone cannot be changed unless a PS changes the setting.
- The following features are not available for extensions in Wireless XDP Parallel Mode while the PS is on a call (however, they are available for the extensions in Wireless XDP Parallel Mode while the wired telephone is on a call):
  - Automatic Callback Busy

  - Executive Busy Override
  - Whisper OHCA (→ 1.7.4.4 Whisper OHCA)
  - CCBS (→ 1.19.1.9 Completion of Calls to Busy Subscriber (CCBS))

- Most of the extension data (e.g., extension number, extension name) of the wired telephone is used for its PS as well. However, the PS has its own extension data for the following:

  - Ring Tone Pattern Table Selection (→ 1.1.3.2 Ring Tone Pattern Selection)
  - Preferred Line Assignment—Incoming (→ 1.4.1.2 Line Preference—Incoming)
  - Preferred Line Assignment—Outgoing (→ 1.5.5.2 Line Preference—Outgoing)
  - Hot Line Setting (→ 1.6.1.7 Hot Line)
  - Transfer Recall Destination for Call Transfer and Call Park
  - Display Language (→ 1.18.4 Display Information)
  - ISDN Bearer Mode (→ 1.19.1.1 Integrated Services Digital Network (ISDN)—SUMMARY)
  - Flexible Button Assignment (→ 1.18.2 Flexible Buttons)

  Note

  To change the setting of the extension data above, the setting for the wired telephone or the PS must be changed individually. When changing the PS setting, use the PS's original extension number (not the main telephone's extension number), if required.

- When the Wireless XDP Parallel Mode has been set, the following extension data for the wired telephone is copied to the PBX extension data for the PS and the extension data remains there even when the Wireless XDP Parallel Mode is cancelled.

  - Call Waiting Setting (→ 1.1.3.3 Call Waiting)
  - FWD/DND Setting (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND))
  - Call Pickup Deny Setting (→ 1.4.1.3 Call Pickup)
  - Executive Busy Override Deny Setting
  - Itemised Billing Code for ARS (→ 1.9.1 Automatic Route Selection (ARS))
  - Transfer Recall Destination for Call Transfer and Call Park
  - CLIP/COLP Number (→ CLIP/COLP Number [606])
  - CLIP/COLP Number Selection
  - CLIR Setting
  - COLR Setting
  - Extension Personal Identification Number (PIN) (→ Extension Personal Identification Number (PIN) [005]) (→ 1.25.1 Extension Personal Identification Number (PIN))
  - COS Programming (→ Class of Service [602]) (→ 2.2.1 Class of Service (COS))
  - User Group (→ User Group [603]) (→ 2.2.2 Group)

- If the Extension Feature Clear is performed, the corresponding extension data for both the wired telephone and the PS will be cleared. (→ 1.25.2 Extension Feature Clear)
- When a call arrives, both the wired telephone and the PS ring. However, in the following cases only the person who made the setting will receive ringing:

  - Automatic Callback Busy
  - Transfer Recall, Hold Recall (→ 1.12.1 Call Hold), and Call Park Recall

Feature Guide References

1.7.1 Automatic Callback Busy (Camp-on)
1.7.2 Executive Busy Override
1.10.9 Parallelled Telephone
1.11.1 Call Transfer
1.12.2 Call Park
1.19.1.2 Calling/Connected Line Identification Presentation (CLIP/COLP)

User Manual References

User Manual

1.7.12 Using Your PS in Parallel with a Wired Telephone (Wireless XDP Parallel Mode)

1.24 Administrative Information Output Features

1.24.1 Station Message Detail Recording (SMDR)

Description

Automatically records detailed information for each extension.

1. **SMDR Output Port**

   The Serial Interface (RS-232C) port can be used to output the SMDR data.
   The following devices can be connected:

   - **Serial Interface (RS-232C) port:** Personal computer (PC), printer, etc.

2. **SMDR Output Data**

   The following data will be recorded and sent to the SMDR output port:

   a) Trunk call information (incoming/outgoing)

   b) Intercom call information (outgoing)

   c) Log-in/Log-out information

   d) PBX error log (→ 2.4.3 Local Alarm Information)

   Memory for SMDR: A specified number of call information can be memorised for each PBX. If more calls are originated or received, the oldest data is overwritten by the newest one.

3. SMDR Format Type and Contents

There are the following three types of output format which can be selected through system programming:

Pattern A: 80 digits without call charge information

```
Date      Time     Ext  CO  Dial Number                    Ring  Duration  ACC Code    CD
(8 digits)(7)      (4)  (2) (25)                           (4)   (8)       (10)        (3)
-------------------------------------------------------------------------------------------
01/02/02 10:03AM  1200  01  <I>12345678901234567890        5'15  00:00'00                NA
01/02/02 10:07AM  1200  01  <I>                            0'05  00:01'05  9876543210
01/02/02 10:15AM  1200  01  1234567890123456                    00:01'05  9876543210  TR
01/02/02 10:30AM  *123  01  1234567890123456                    00:01'05  9876543210


01/02/02 01:07PM  1234  01  <I>ABC COMPANY12345678   0'05  00:01'05  9876543210
01/02/02 01:07PM  1234  01  <D>CDE9876<I>Q COMPANY   0'05  00:01'05  9876543210
01/02/02 01:07PM  1234  01  ABC COMPANY12345678           00:01'05
01/02/02 01:07PM  1234  01  123.....                     00:01'05
01/02/02 01:07PM  1234  01  123456XX                     00:12'05  98765

01/02/02 08:33AM  1234      In the office
01/02/02 01:07PM  1234      LOG IN
01/02/02 03:35PM  1234      LOG OUT
01/02/02 03:45PM  1234      EXT12355

    :        :        :   :       :                      :     :         :         :
   (1)      (2)      (3) (4)     (5)                    (6)   (7)       (8)       (9)
```

Pattern B: 80 digits with call charge information

```
Date      Time     Ext  CO  Dial Number                    Duration  Cost        ACC Code    CD
(8 digits)(7)      (4)  (2) (20)                           (8)       (8+2)       (10)        (3)
-------------------------------------------------------------------------------------------
01/02/02 10:03AM  1210  01  <I>                                                               NA
01/02/02 10:07AM  2005  01  12345678901234567890  00:00'05  00560.00EU  9876543210

    :        :        :   :       :                  :         :           :         :
   (1)      (2)      (3) (4)     (5)                (7)       (10)        (8)       (9)
```

Pattern C: 120 digits

```
Date      Time     Ext  CO  Dial Number                                  Ring  Duration  Cost        ACC Code    CD
(8 digits)(7)      (4)  (4) (50)                                         (4)   (8)       (8+3)       (10)        (3)
-------------------------------------------------------------------------------------------------------------------
01/02/02 10:03AM  1230  0001 12345678901234567890123456789012345678901234567890        00:00'05  00560.00EUR 9876543210  TR
01/02/02 10:07AM  1230  0001 <I>ABC COMPANY123456789012345              0'05  00:00'05              9876543210  TR

    :        :        :   :       :                                      :     :         :           :         :
   (1)      (2)      (3) (4)     (5)                                    (6)   (7)       (10)        (8)       (9)
```

[Explanation]
The following table explains the SMDR contents which are based on the numbers in the previous pattern examples. For the programmable items, refer to the following [Programmable Items].

Number in the Pattern	Item	Description
(1)	**Date**	Shows the date of the call.
(2)	**Time**	Shows the end time of a call as Hour/Minute/AM or PM.
(3)	**Ext (Extension)**	Shows the extension number, floating extension number, etc., which was engaged in the call. Also shows the following codes: **Dxxx:** Outgoing trunk call from a doorphone (xxx=doorphone number) (→ 1.15.1 Doorphone Call) **Txxx:** Outgoing trunk call by TIE line service (xxx=trunk group number) ***xxx:** Verified call (xxx=verified code) (→ 1.8.6 Verified Code Entry)
(4)	**CO (Trunk)**	Shows the trunk number used for the call. For patterns A and B, "00" will be shown for trunk numbers over hundred.
(5)	**Dial Number**	**[Trunk Call]** **Outgoing Trunk Call** Shows the dialled telephone number. Valid digits are as follows: **0 through 9, ✳ , #** **P:** Pause **F:** EFA signal **=:** A Host PBX Access code (→ 1.5.4.7 Host PBX Access Code (Access Code to the Telephone Company from a Host PBX)) **. (dot):** Secret dialling **X:** Privacy dial **–:** Transferred call If the transfer destination extension enters some digits, the entered digits will be added after "–". **Incoming Trunk Call** Shows <I> + the caller's identification name/number. It is also possible to show the DDI/DID/MSN call information. In this case, <D> + DDI/DID/MSN name/number is added before <I> .
		[Outgoing Intercom Call] Shows the dialled extension number followed by "EXT".
		[Log-in/Log-out] Shows the log-in or log-out status.
(6)	**Ring**	Shows the ring duration before answering a call in Minutes/Seconds.

Number in the Pattern	Item	Description
(7)	**Duration**	Shows the duration of the trunk call in Hours/Minutes/Seconds.
(8)	**Acc Code (Account Code)**	Shows the account code appended to the call. (→ 1.5.4.3 Account Code Entry)
(9)	**CD (Condition Code)**	Shows other call information with the following codes: **CL:** Collect call **TR:** Transfer **FW:** FWD to trunk **D0:** Call using DISA or TIE line service **RM:** Remote maintenance (modem) (→ 2.3.1 PC Programming) **NA:** Not answered call
(10)	**Cost**	Shows the call charge.

[Programmable Items]

Item	Description
Outgoing trunk call	Controls whether the outgoing trunk calls are shown. This setting is common throughout the PBX (→ SMDR Outgoing Call Printing [804]). COS programming is also required.
Incoming trunk call	Controls whether the incoming trunk calls are shown (→ SMDR Incoming Call Printing [805]).
Outgoing intercom call	Controls whether the outgoing intercom calls are recorded.
Log-in/Log-out status	Controls whether the log-in/log-out status is recorded.
ARS dial	Controls whether the user-dialled number or the modified number is shown. The Host PBX Access code ("=" followed by the access code) can be shown (as supplementary information) only when the modified number is selected in this setting. (→ 1.9.1 Automatic Route Selection (ARS))
Caller's identification	Controls whether the caller's identification number, name, number and name, or nothing is shown. If "none" is selected, <I> will not be shown.
DID/DDI number	Controls whether the DID/DDI number, name, number and name, or nothing is shown. If "none" is selected, <D> will not be shown.

Item	Description
Secret dialling	Controls secret dialling. If enabled, the dialled number will be shown as dots. This setting is effective only when the modified number is selected in ARS dial setting above. If the user-dialled number is selected in ARS dial setting, the dialled number will be shown as dots regardless of this setting.
Privacy dial	Enables or disables privacy dial. If enabled, the last four digits of the dialled telephone number and any additional digits after connection will be shown as "X". (e.g., 123-456-XXXX)
Date order	The date order is changeable: month/day/year, day/month/year, year/month/day, year/day/month.

Conditions

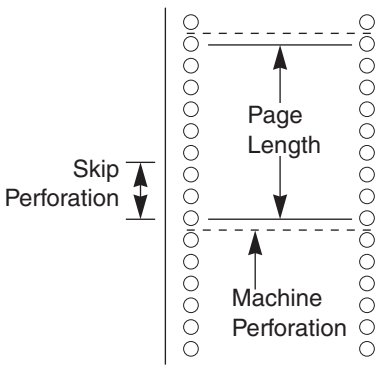
- **SMDR Format**
  The following SMDR format can be set through system programming in order to match the paper size being used in the printer:

  a) **Page Length:** determines the number of lines per page (→ SMDR Page Length [802]).

  b) **Skip Perforation:** determines the number of lines to be skipped at the end of every page (→ SMDR Skip Perforation [803]).

  The page length should be at least four lines longer than the skip perforation length.

  Explanation:



- The SMDR data is not deleted even if the PBX is reset.
- If the PBX is reset during a conversation, the call will not be recorded on the SMDR.
- The following calls are regarded as two separated calls in the SMDR:
  - Calls before and after the flash/recall/EFA signal is manually sent during a conversation
  - Trunk-to-trunk calls by Call Transfer, FWD or DISA (recording each as "incoming call" and "outgoing call")
- The PBX waits for a preprogrammed time period (→ Call Duration Count Starting Time for LCOT [208]) between the end of dialling and start of the SMDR timer for outgoing trunk calls. When the PBX has sent out all dialled digits to the telephone company and this timer expires, the PBX starts counting the call. A display PT shows the elapsed time of the call.

The starting time and the total duration of the call are recorded on SMDR.
If the reverse signal detection has been set (→ 1.5.4.5 Reverse Circuit), the PBX will start counting the call after detecting the reverse signal from the telephone company regardless of the above timer.

- **Serial Interface (RS-232C) Parameters**
  The following communication parameters can be assigned for the Serial Interface (RS-232C) port (→ RS-232C Parameter—New Line Code [800], RS-232C Parameter—Baud Rate [800], RS-232C Parameter—Word Length [800], RS-232C Parameter—Parity Bit [800], RS-232C Parameter—Stop Bit Length [800]).
  - **a)** **New Line Code:** Select the code for the PC or printer. If the PC or printer automatically feeds lines with carriage return, select "CR." If not, select "CR+LF."
  - **b)** **Baud Rate:** A baud rate code indicates the data transmission speed from the PBX to the PC or printer.
  - **c)** **Word Length:** A word length code indicates how many bits compose a character.
  - **d)** **Parity Bit:** A parity code indicates what type of parity is used to detect an error in the string of bits composing a character. Make an appropriate selection depending on the requirements of the PC or printer.
  - **e)** **Stop Bit Length:** A stop bit code indicates the end of a bit string which composes a character. Select an appropriate value depending on the requirements of the PC or printer.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.11.1 Connection of Peripherals

<KX-TDA30>

6.10.1 Connection of Peripherals

Feature Guide References

1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)

1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service

1.2.2.6 Log-in/Log-out

1.11.1 Call Transfer

1.15.6 Direct Inward System Access (DISA)

1.27.1 TIE Line Service

1.24.2 Charge Meter

Description

The PBX receives a call charge signal during or after a conversation with an outside party. The call charge information is shown on the telephone display and the SMDR.

1. Call Charge Signal Service

Call charge signal service from the telephone company depends on the trunk being used.

Trunk	Service
Analogue trunk	Pay Tone (KX-TDA100/KX-TDA200 only)*
ISDN line	Advice of Charge (AOC) (→ 1.19.1.3 Advice of Charge (AOC))
E1 line	Meter Pulse (KX-TDA100/KX-TDA200 only)

*: When the pay tone service is used, pay tone type (12 kHz/16 kHz) should be selected (→ Pay Tone Signal Type [491] (KX-TDA100/KX-TDA200 only)).

2. Call Charge Display

– Up to 7 digits (e.g., 88888.75)
– The decimal point position (the number of significant decimal digits) for currency is programmable. (→ Decimal Point Position for Currency [130])
– Up to 3 currency characters are programmable. (e.g., EUR or € for Euro) (→ Currency [131]).
– The currency position either front or back of the call charge is programmable. (e.g., € 45.12 or 45.12 €)

3. Margin/Tax Rate Assignment

It is possible to add a margin (→ Charge Margin [010]) and the tax (→ Charge Tax [011]) to the call charges. The call charge rate per meter indication is programmable on a trunk group basis (→ Charge Rate per Unit [012]).
[Calculation Method]
The margin or tax rate must be xx.xx % (whole number part: two digits, decimal fraction: two digits). A calculation method depends on the information which is sent from the telephone company: meter indication or charge.

a) Call charge with tax and margin in meter indication:

$$\frac{[\text{Meter indication received from the telephone company}] \times [\text{Call Charge Rate}]}{[1 - \text{Tax Rate}] \times [1 - \text{ Margin Rate}]}$$

b) Call charge with tax and margin in charge:

$$\frac{[\text{Charge received from the telephone company}] \times [\text{Call Charge Rate}]}{[1 - \text{Tax Rate}] \times [1 - \text{Margin Rate}]}$$

The calculation result is rounded up to the least significant decimal digit.

4. Total Call Charge

– A PT user can show the total call charges on the display.
– The call charge is totalled on an extension, trunk, or verified code basis.
– The call charge on a verified code is not totalled for each extension.

5. Budget Management

It is possible to limit telephone usage to a preprogrammed budget on each extension or verified code. For example, an extension in a rented office has a prepaid limit for the telephone usage. If the amount of the call charge reaches the limit, the extension user cannot make further trunk calls. An extension assigned as the manager may increase the limit or clear the previous call charge (→ 1.8.2 Budget Management).

6. Call Charge Management

A manager can use the following call charge management features in manager programming mode:

a) Call Charge Fee Reference and Clear, for Each Extension/Verified Code
b) Call Charge Fee Reference for Each Trunk
c) Call Charge Rate Set for Each Trunk Group
d) All Clear
e) Call Charge Fee Print Out
f) Budget Management for Each Extension/Verified Code.

[Examples of Call Charge Fee Reference]

```
*******************************************************
*   Charge Meter Print Out - Total & All CO     *
*******************************************************
Total Charge:  €00175.95

CO Line
  001:  €00194.00      002:  €00073.00      003:  €00161.00      004:  €00033.00
```

```
*******************************************************
*   Charge Meter Print Out - All Extensions     *
*******************************************************
Extension & Verified Code
 *775:  €00194.00      *102:  €00073.00      *776:  €00161.00  ←— (Verified Code)

  104:  €00194.00      105:  €00073.00      106:  €00161.00      107:  €00033.00
```

Note: *XXX: Call Charge Fee Reference for a verified code (XXX = verified code)

Conditions

[General]

- **Call Charge Fee Reference by Charge Reference Button**
  A display telephone user can check the total call charge for his own extension using the Charge Reference button. A flexible button can be customised as the Charge Reference button.

[Pay Tone Service (KX-TDA100/KX-TDA200 only)]

- It is possible to select whether the PBX starts counting the call charge from when the PBX detects the answer signal from the telephone company.
- It is possible to enable the PBX to send the flash/recall signal to the telephone company after a conversation (in on-hook status) to receive the call charge information.

Feature Guide References

1.8.6 Verified Code Entry

1.18.2 Flexible Buttons

User Manual References

User Manual

3.2.2 Manager Programming

1.25 Extension Controlling Features

1.25.1 Extension Personal Identification Number (PIN)

Description

Each extension user can have his own PIN through system programming (→ Extension Personal Identification Number (PIN) [005]) or personal programming (Extension PIN [Personal Identification Number]) to set features or access his own telephone remotely.
The following features cannot be used without the PIN:

a) Live Call Screening (LCS)* (→ 1.22.3 Voice Mail DPT (Digital) Integration)

b) Incoming and Outgoing Call Log Display Lock (→ 1.16.2 Incoming Call Log), Personal Speed Dialling Display Lock (→ 1.6.1.5 Speed Dialling—Personal/System)

c) Walking Extension (→ 1.25.3 Walking Extension)

d) Extension Lock (→ 1.8.3 Extension Lock)

e) Remote Extension Control (→ 1.25.5 Remote Extension Control by User) and Walking COS (→ 1.8.5 Walking COS)

f) Remote Extension Control and Walking COS using DISA (→ 1.15.6 Direct Inward System Access (DISA))

*: If an extension user has assigned an extension PIN, this feature cannot be used without the PIN.

Conditions

WARNING
When a third party discovers the personal identification number (PIN) (verified code PIN/ extension PIN) of the PBX, there will be a risk that they will make fraudulent phone calls using the telephone line, and the cost will be charged to your own account. In order to avoid this problem, we strongly recommend the following points:

a) Carefully maintain the secrecy of the PIN.

b) Specify a complicated PIN as long and random as you can make it.

c) Change the PIN frequently.

- **Extension PIN Lock**
  If the wrong PIN is entered incorrectly three times, the line will be disconnected. If the incorrect PIN is entered a preprogrammed number of times successively, the PIN against the extension will be locked. Only an extension assigned as the manager can unlock it. In this case, the PIN will be unlocked and cleared.
  This feature is also known as Station Password Lock.
- **Remote Extension PIN Clear**
  If an extension user forgets his PIN, a manager can clear the PIN. Then the extension user can assign a new PIN.
- **Extension PIN Display**
  It is possible to select whether to show the extension PIN on the display through system programming. By default, it is shown as dots.

User Manual References

User Manual

3.1.2 Settings on the Programming Mode

3.2.2 Manager Programming

1.25.2 Extension Feature Clear

Description

Extension users can clear all the following features set on their own telephone at once:

Features	After Setting
Absent Message	**Off**
BGM	**Off**
FWD*/DND*	**Off**
Call Pickup Deny	**Allow**
Call Waiting*	**Disable** (In Canada, the default setting is "Enable" [Call Waiting tone].)
Data Line Security	**Off**
Executive Busy Override Deny	**Allow**
Log-in/Log-out	**Log-in**
Message Waiting	**All messages left by other extensions will be cleared.**
Paging Deny	**Allow**
Parallelled Telephone	**Paired SLT will ring.**
Hot Line*	**Off**
Timed Reminder	**Cleared**

Note: The features with "*" can be programmed not to be cancelled by this feature.

This feature is also known as Station Programme Clear.

Conditions

- Extension Lock (→ 1.8.3 Extension Lock) and the extension personal identification number (PIN) (→ 1.25.1 Extension Personal Identification Number (PIN)) will not be cleared by this feature.
- **For Users in Canada only**
  If dial tone 2 is heard after Extension Feature Clear:
  After performing Extension Feature Clear, Call Waiting will be enabled if "`Extension Clear: Call Waiting`" is set to "`Clear`" through system programming. In this case, dial tone 2 will be heard when going off-hook. (→ 1.26.1 Dial Tone)

User Manual References

User Manual

1.7.13 Clearing Features Set at Your Extension (Extension Feature Clear)

1.25.3 Walking Extension

Description

It is possible to use any extension and have your extension settings available to you. Settings such as extension number, one-touch dialling memory, and COS are all available to you at the new location.
[Example] This feature is useful when;

- Moving location
- There is no specific desk for your use.

This feature is also known as Walking Station.

Conditions

- This feature is available by switching the telephone between PT and PT, SLT and SLT, PT and SLT. Moving across the tenants is also possible.
- The incoming calls to your extension will also reach you at your new location.
- An extension personal identification number (PIN) is required to use this feature. (→ 1.25.1 Extension Personal Identification Number (PIN))
- If a DSS Console is connected to a PT and the DSS Console is continuously used with the PT after the Walking Extension has been done, the new extension number of the PT must be assigned as the paired extension through system programming (→ Console Paired Telephone [007]).

User Manual References

User Manual

1.10.1 Using the Same Settings as Your Previous Extension (Walking Extension)

1.25.4 Timed Reminder

Description

Each extension user can set an alarm for any time as a wake-up call or reminder. This feature can be programmed to activate only once or daily. If the user goes off-hook during the alarm, he hears special dial tone. If a voice message is used, he hears a prerecorded voice message.

Conditions

- Be sure that the PBX clock works.
- Setting a new time clears the previous time.
- **Programmable Time**
  The Alarm Ringing Duration time, the number of alarm repeat times, and intervals are programmable through system programming.
- **To use the voice message feature:**
  The MSG card is required and the message must be recorded. An extension assigned as the manager can record messages. (→ 1.15.5 Outgoing Message (OGM)) The different message can be assigned for each time mode (day/lunch/break/night) (→ 2.2.4 Time Service).

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.6.5 MSG4 Card

<KX-TDA30>

6.5.4 MSG2 Card

User Manual References

User Manual

1.7.1 Setting the Alarm (Timed Reminder)

1.25.5 Remote Extension Control by User

Description

A user can remotely change the feature settings (e.g., FWD) and time service mode for his own extension from inside or outside the PBX using DISA (→ 1.15.6 Direct Inward System Access (DISA)). This feature can be used only on the extensions which are allowed to be controlled remotely on a COS basis.

The following features are available:

a) FWD/DND (→ 1.3.1 Call Forwarding (FWD)/Do Not Disturb (DND))

b) Log-in/Log-out (→ 1.2.2.6 Log-in/Log-out)

c) Absent Message (→ 1.17.2 Absent Message)

d) Extension Lock (→ 1.8.3 Extension Lock)

e) Time Service Switching Mode (→ 2.2.4 Time Service)

This feature is also known as Remote Station Control by User.

Conditions

- **Extension PIN**
  An extension personal identification number (PIN) (→ Extension Personal Identification Number (PIN) [005]) is required to use this feature. (→ 1.25.1 Extension Personal Identification Number (PIN)) If the wrong PIN is entered three times, the line will be disconnected.
- Remote operation is not available for the extensions in the Internal Call Block mode. (→ 1.1.2.2 Internal Call Block)

User Manual References

User Manual

1.2.9 Setting Your Telephone from Another Extension or through DISA (Remote Setting)

1.26 Audible Tone Features

1.26.1 Dial Tone

Description

The following dial tones inform extensions about features activated on the extensions:
Each dial tone type has two frequencies (e.g., dial tone 1A and dial tone 1B).

Type	Description
Tone 1A/1B	A normal dial tone is heard when: **a)** No features listed for dial tones 2 through 4 has been set, or **b)** ARS is used.
Tone 2A/2B	Any one of the features below is set. • Absent Message • BGM • FWD • Call Pickup Deny • Call Waiting • DND • Extension Lock • Executive Busy Override Deny • Hot Line • Timed Reminder
Tone 3A/3B	While a called PS is being searched, or any one of the features below is performed. • Account Code Entry • Consultation Hold • Answering a Timed Reminder call with no message
Tone 4A/4B	Messages are waiting for the extension.

Conditions

- **Dial Tone Type A/B**
  It is possible to select dial tone type A or B for dial tones 1 through 4. If "`Type A`" is selected, all dial tones 1 through 4 will become dial tone type A.
  The dial tone type for the ARS feature can be selected separately. If "`Type A`" is selected for the ARS, dial tone 1A will be heard. If "`Type B`" is selected, dial tone 1B will be heard.
- **Distinctive Dial tone**
  Distinctive dial tone mode can be disabled. In this case, dial tone 1 will be sent.
- **Dial Tone Patterns**
  All dial tone patterns have a default (→ 4.2.1 Tones/Ring Tones).
- Only dial tone 1 is sent to the extensions in the VM (DPT/DTMF) group. (→ 1.22.1 Voice Mail (VM) Group)

1.26.2 Confirmation Tone

Description

At the end of feature operations, the PBX confirms the success of the operation by sending a confirmation tone to extension users.

Type	Description
Tone 1	**a)** Sent when the setting is accepted. **b)** Sent when a call is received in voice-calling mode (Alternate Receiving—Ring/Voice). The caller's voice will be heard after the tone.
Tone 2	**a)** Sent from an external paging device or an extension before being paged. **b)** Sent when a call is received in Hands-free Answerback mode.
Tone 3-1	**a)** Sent before a conversation is established when using the Paging feature. **b)** Sent when a conversation is established with the extension in the following modes after the call making operation: • Hands-free Answerback mode • Voice-calling mode (Alternate Receiving—Ring/Voice)
Tone 3-2	Sent just before a conversation is established when accessing the following features by the feature numbers: • Call Park Retrieve • Call Pickup • Hold Retrieve • Paging Answer • TAFAS
Tone 4-1	Sent when moving from a two-party call to a three-party call. (e.g., Executive Busy Override, Conference, Privacy Release, Two-way Record.)
Tone 4-2	Sent when moving from a three-party call to a two-party call. (e.g., Executive Busy Override, Conference, Privacy Release, Two-way Record.)
Tone 5	Sent when a call is placed on hold (including Consultation Hold).

Conditions

- **Confirmation Tone Patterns**
  All confirmation tone patterns have a default (→ 4.2.1 Tones/Ring Tones).
- It is possible to eliminate each tone.

1.27 Networking Features

1.27.1 TIE Line Service

Description

A TIE line is a privately leased communication line between two or more PBXs, which provides cost effective communications between company members at different locations. The TIE lines can be used to call through the PBX to reach another switching system (PBX or telephone company). By using TIE lines, the PBX can support not only communications with the public network but with other company members in the private network where the PBX is included.

Interface

The following interfaces can be used to establish a private network:

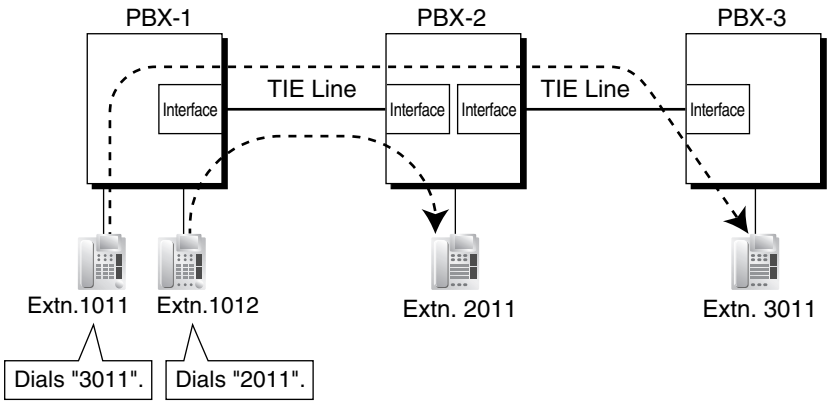
Interface	Network Type
E & M	Analogue
T1 (TIE [E & M])	Digital (64 kbps × 24 channels)
E1 (E & M)	Digital (64 kbps × 30 channels)
BRI/PRI (QSIG)	Digital (ISDN 2B+D/30B+D/23B+D)
VoIP	Internet Protocol (IP)

A trunk which is used for a private network should be assigned "`Private`" as the networking type. (→ 1.1.1.1 Incoming Trunk Call Features—SUMMARY)

Explanation

1. **Making a TIE Line Call**
   One of the following two methods can be used to make a TIE line call.
   a) **Extension Number Method (Access without PBX Code)**
      Dial the [Extension Number] only.

[Example]



Explanation:
To use this method, it is necessary to change the first one or two digits of extension numbers of either PBX (e.g., 10XX for PBX-1, 20XX for PBX-2) to avoid having the same extension number.

Case 1:

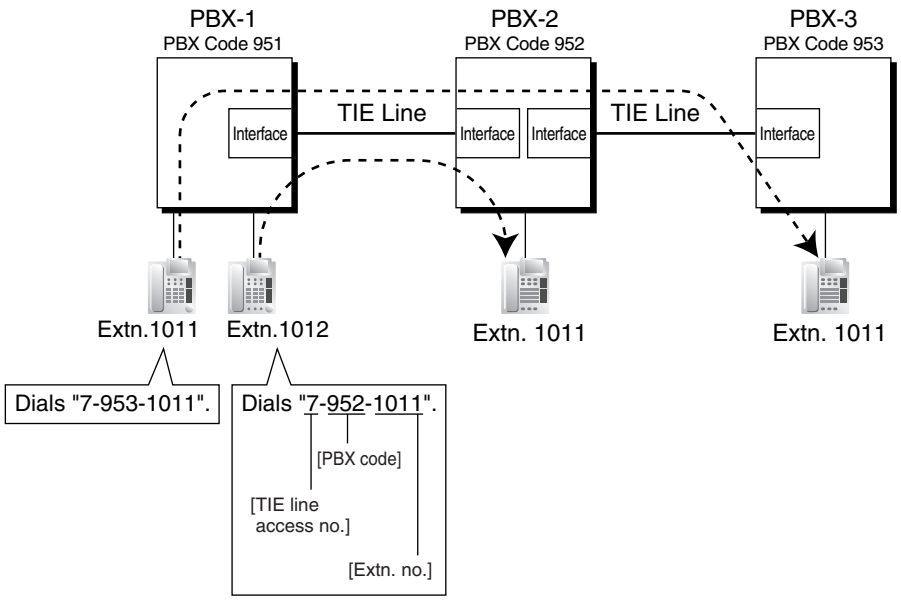
Extension 1012 of PBX-1 dials extension number "2011".
→ Extension 1012 of PBX-1 is connected to extension "2011" of PBX-2.

Case 2:

Extension 1011 of PBX-1 dials extension number "3011".
→ Extension 1011 of PBX-1 is connected to extension "3011" of PBX-3.

b) PBX Code Method (Access with PBX Code)

Dial the [TIE Line Access Number] + [PBX Code] + [Extension Number].
[Example]



Explanation:
To use this method, it is necessary to know each PBX code in order to identify the location of an extension.

Case 1:

Extension 1012 of PBX-1 dials TIE line access number "7", PBX code "952", and extension number "1011".
→ Extension 1012 of PBX-1 is connected to extension "1011" of PBX-2.

Case 2:

Extension 1011 of PBX-1 dials TIE line access number "7", PBX code "953", and extension number "1011".
→ Extension 1011 of PBX-1 is connected to extension "1011" of PBX-3.

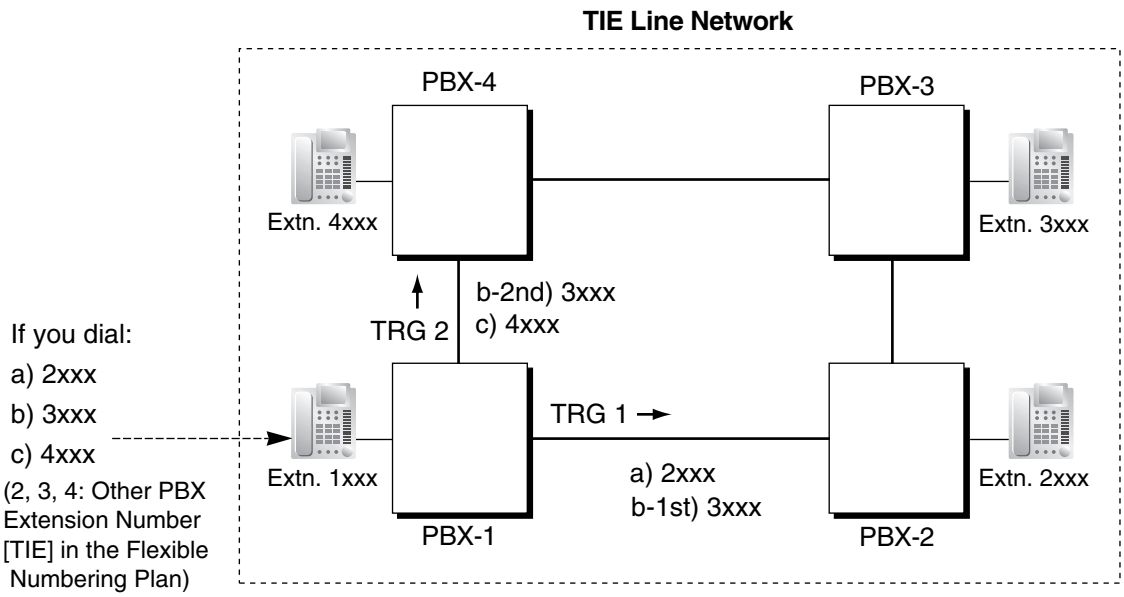
2. TIE Line Routing and Modification Table

2.1 To Make a TIE Line Call

The TIE Line Routing and Modification Table is referenced by the PBX to identify the trunk route when an extension user makes a TIE line call.
It is necessary to make unified tables with all PBXs in the TIE line network.
The routing pattern appropriate for each call is decided by the dialled number.
There are two system programmes for the tables:
TIE Line Routing Table: used to assign the leading numbers (PBX code or extension number) and trunk group hunt sequence.
TIE Modify Removed Number of Digits/Added Number: used to remove digits from and add a number to the dialled number of the TIE line call. This modification may be needed depending on the TIE line network configuration.

[Programming Examples]
Your PBX is PBX-1 and there are four PBXs in your TIE line network. To identify the trunk route as illustrated, you should make the following tables.

a) Extension Number Method (Access without PBX Code)



[TIE Line Routing and Modification Table of PBX-1]

Location No.	Leading No.	Priority 1			Priority 2			. .
		Trunk Group	Dial Modification		Trunk Group	Dial Modification		. .
			Removed No. of Digits	Added No.		Removed No. of Digits	Added No.	. .
01	2	1	0					. .
02	3	1	0		2	0		. .
03	4	2	0					. .
:	:	:	:	:	:	:	:	:

Explanation:

Location 01: The hunt sequence by dialling [2XXX]:

The 1st route—trunk group (TRG) 1 ➡ Sending no. to PBX-2: 2XXX

Location 02: The hunt sequence by dialling [3XXX]:

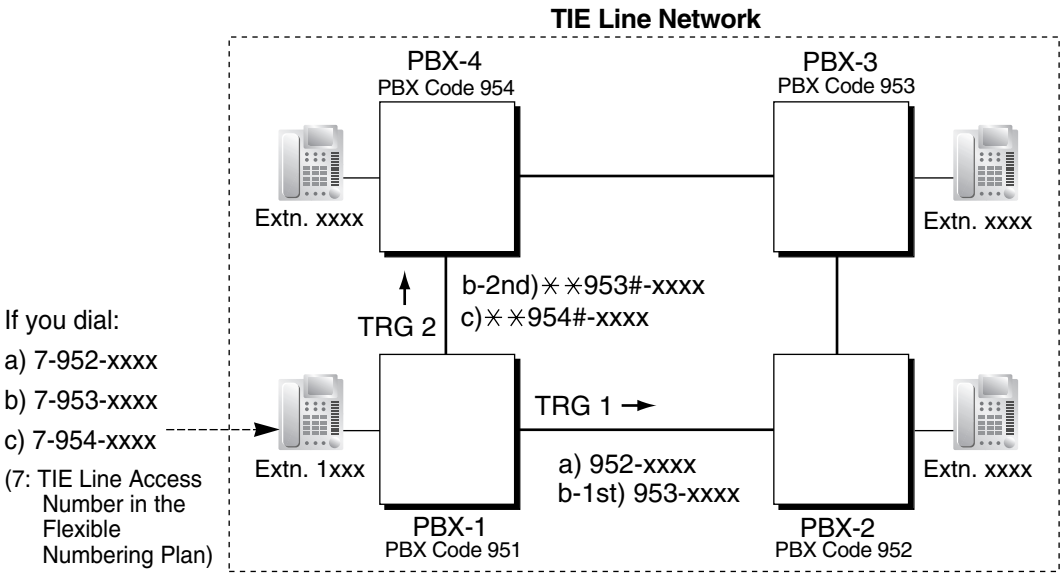
The 1st route—trunk group (TRG) 1 ➡ Sending no. to PBX-2: 3XXX

The 2nd route—trunk group (TRG) 2 ➡ Sending no. to PBX-4: 3XXX

Location 03: The hunt sequence by dialling [4XXX]:

The 1st route—trunk group (TRG) 2 ➡ Sending no. to PBX-4: 4XXX

b) PBX Code Method (Access with PBX Code)



[TIE Line Routing and Modification Table of PBX-1]

Location No.	Leading No.	Priority 1			Priority 2			. .
		Trunk Group	Dial Modification		Trunk Group	Dial Modification		. .
			Removed No. of Digits	Added No.		Removed No. of Digits	Added No.	. .
01	952	1	0					. .
02	953	1	0		2	3	✳✳953#	. .
03	954	2	3	✳✳954#				. .
:	:	:	:	:	:	:	:	:

Explanation:

Location 01: The hunt sequence by dialling [7+PBX Code 952+XXXX]:

The 1st route — trunk group (TRG) 1

➨ Sending no. to PBX-2: 952–XXXX

Location 02: The hunt sequence by dialling [7+PBX Code 953+XXXX]:

The 1st route — trunk group (TRG) 1

➨ Sending no. to PBX-2: 953–XXXX

The 2nd route — trunk group (TRG) 2

➨ Sending no. to PBX-4: ✳✳953#–XXXX

Location 03: The hunt sequence by dialling [7+PBX Code 954+XXXX]:

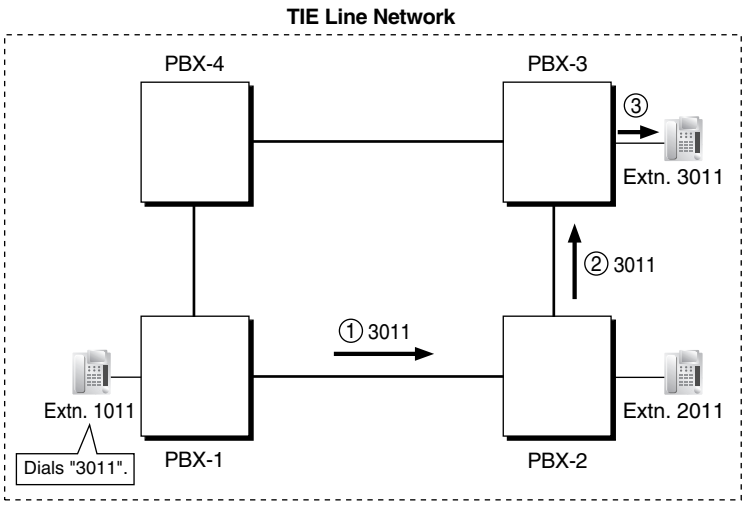
The 1st route — trunk group (TRG) 2

➨ Sending no. to PBX-4: ✳✳954#–XXXX

2.2 To Receive a TIE Line Call

a) Extension Number Method (Access without PBX Code)

[Example]



① A TIE line call is sent to PBX-2 from PBX-1. If the number sent from PBX-1 is an extension number of PBX-2 (e.g., 2011), the call will be received at extension "2011". If not, PBX-2 checks the number in the TIE Line Routing and Modified Table of PBX-2.

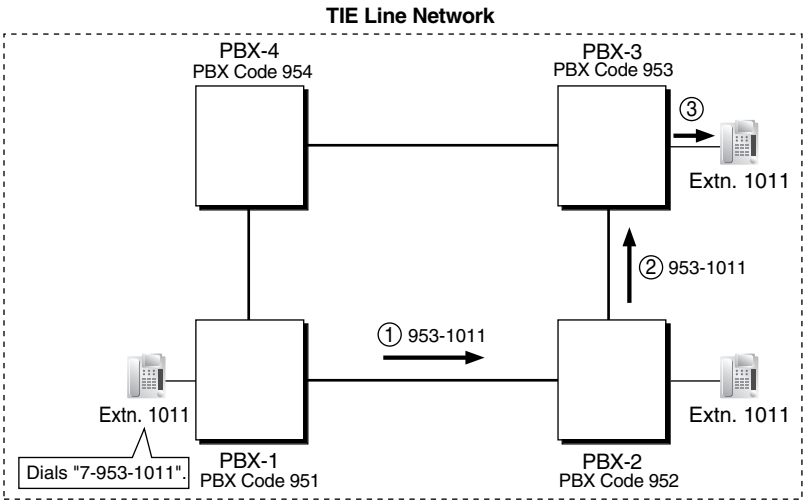
② If the match is found in the table, the call will be modified according to the table and send to the corresponding PBX (PBX-3).

③ The number sent from PBX-2 "3011" is an extension number of PBX-3. The call is received at extension "3011".

Note

When a TIE line call is sent to a PBX from another PBX, first the PBX modifies the number sent to the PBX according to the assignment for each trunk port of the PBX: the removed number of digits from and/or added number to the number sent to the PBX is determined by the assignment. Then the PBX starts to check the number whether the number is an extension number of the PBX.

b) PBX Code Method (Access with PBX Code)
[Example]



① A TIE line call is sent to PBX-2 from PBX-1. If the number sent from PBX-1 has the PBX code of PBX-2 "952", the call will be received at the corresponding extension of PBX-2 (e.g., 1011of PBX-2). If not, PBX-2 checks the number in the TIE Line Routing and Modified Table of PBX-2.

② If the match is found in the table, the call will be modified according to the table and send to the corresponding PBX (PBX-3).

③ The number sent from PBX-2 "953-1011" has the PBX code of PBX-3 "953". The call is received at extension "1011" of PBX-3.

Note

When a TIE line call is sent to a PBX from another PBX, first the PBX modifies the number sent to the PBX according to the assignment for each trunk port of the PBX: the removed number of digits from and/or added number to the number sent to the PBX is determined by the assignment. Then the PBX starts to check the number whether the number has the PBX code of the PBX.

3. TIE Line and Trunk Connection

To connect the TIE line with the trunk, the following patterns are available:

1) Trunk-to-TIE Access

2) TIE-to-Trunk Access

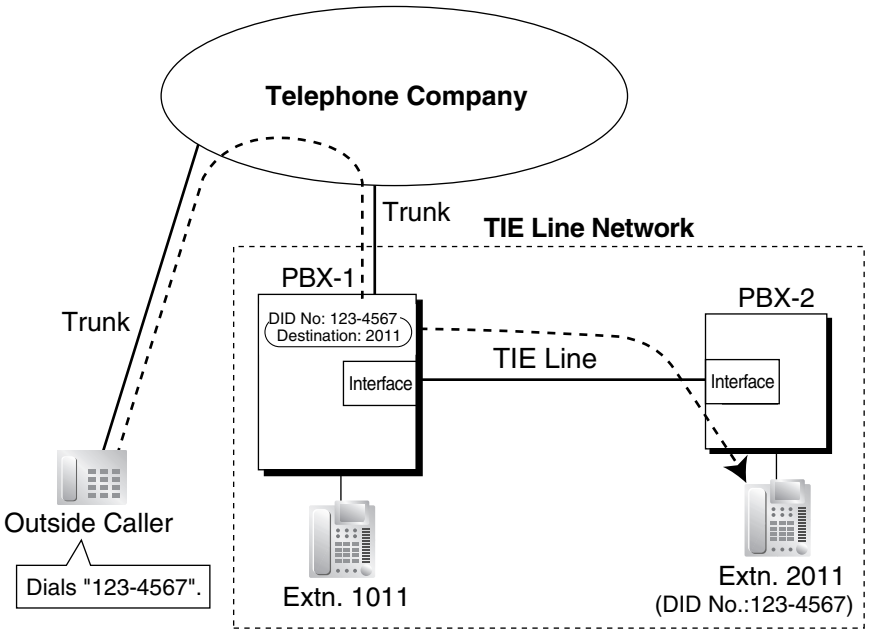
3) Trunk-to-TIE-to-Trunk Access

3.1 Trunk-to-TIE Access

It is possible to assign an extension of another PBX as the destination of incoming trunk calls to the own PBX.

a) Incoming Trunk Call Destination Assignment

[Example]

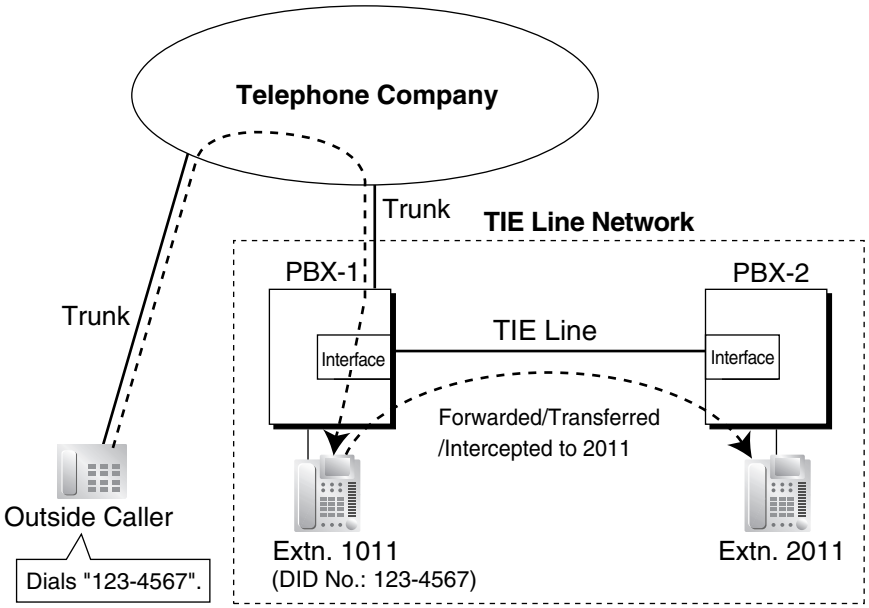


Explanation:

An outside caller dials the DID number "123-4567". The call is sent to extension "2011" of PBX-2 through the TIE line according to the assignment of the DID call destination (→ DID Destination [453]) of PBX-1. (→ 1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI))

b) FWD/Call Transfer/Intercept Routing to the TIE Line

[Example]



Explanation:
An outside caller dials the DID number "123-4567". The call reaches the destination (extension 1011 of PBX-1), and the call is forwarded, transferred, or intercepted to extension "2011" of PBX-2 through the TIE line.

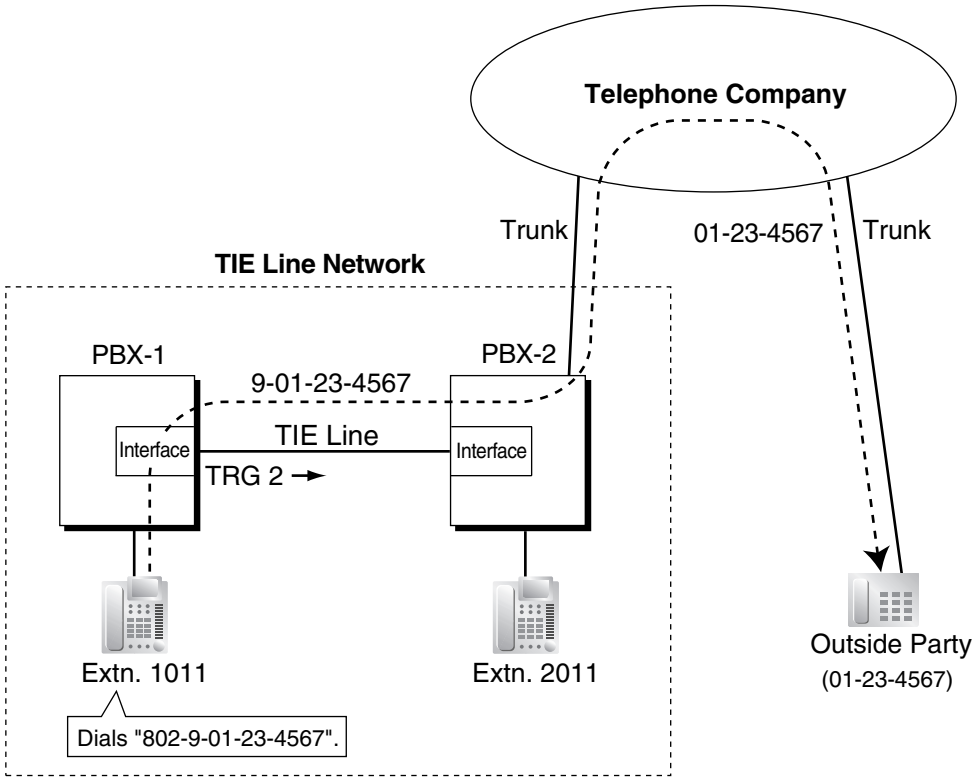
3.2 TIE-to-Trunk Access

The PBX sends TIE line calls to the trunks of another PBX through the TIE lines.

a) Trunk Call through Other PBXs—by Making a TIE Line Call Method

[Example]

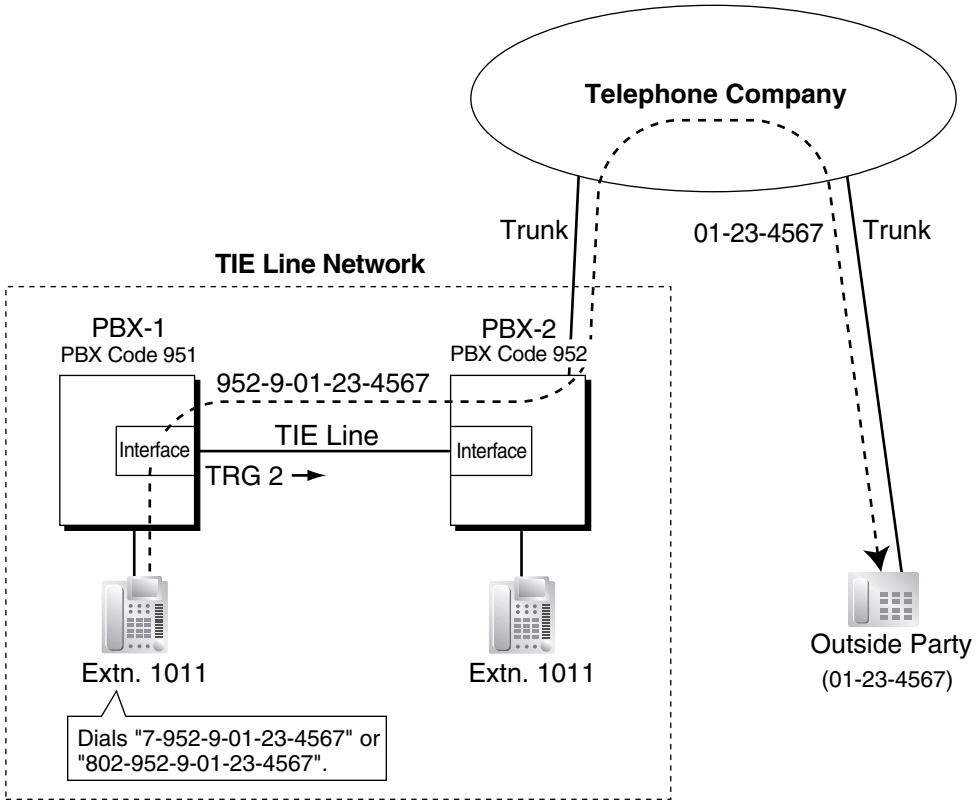
<Extension Number Method (Access without PBX Code)>



Explanation:

1. Extension 1011 of PBX-1 dials the Trunk Group Access number of PBX-1 "8", trunk group number "02" (TRG2), Idle Line Access number of PBX-2 "9", and telephone number "01-23-4567".
2. PBX-1 sends the call to PBX-2 through the trunk group (TRG) 2 (TIE line).
3. PBX-2 sends the call to the outside party "01-23-4567".

<PBX Code Method (Access with PBX Code)>



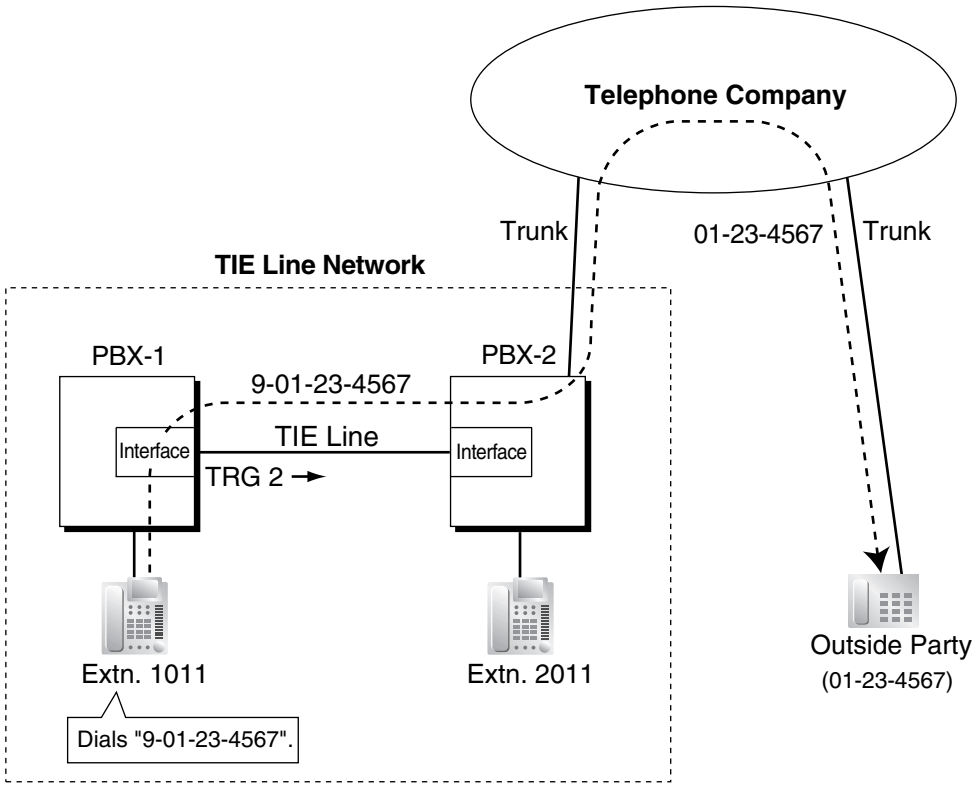
Explanation:

1. Extension 1011 of PBX-1 dials the TIE line access number "7", PBX code "952", Idle Line Access number of PBX-2 "9", and telephone number "01-23-4567"; or dials the Trunk Group Access number of PBX-1 "8", trunk group number "02" (TRG2), PBX code "952", Idle Line Access number of PBX-2 "9", and telephone number "01-23-4567".
2. The call is connected to the outside party "01-23-4567" through PBX-2 which has PBX code "952".

b) Trunk Call through Other PBXs—by the ARS feature

[Example]

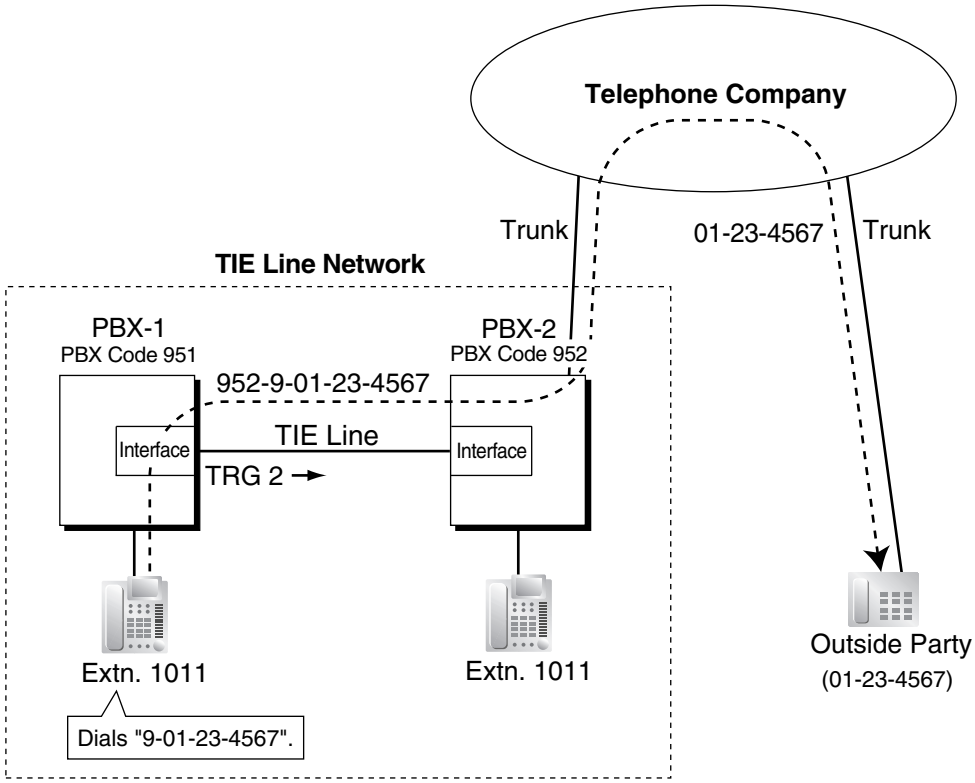
<Extension Number Method (Access without PBX Code)>



Explanation:

1. Extension 1011 of PBX-1 dials the Idle Line Access number of PBX-1 "9" and telephone number "01-23-4567".
2. PBX-1 modifies the call (adds the Idle Line Access number of PBX-2 "9") and sends the call to PBX-2 through the TIE line (trunk group [TRG] 2) according to the ARS programming of PBX-1.
3. PBX-2 sends the call to the outside party "01-23-4567".

<PBX Code Method (Access with PBX Code)>



Explanation:

1. Extension 1011 of PBX-1 dials the Idle Line Access number of PBX-1 "9" and telephone number "01-23-4567".
2. PBX-1 modifies the call (adds "952" and the Idle Line Access number of PBX-2 "9") and sends the call to PBX-2 which has PBX code "952" through the TIE line (trunk group [TRG] 2) according to the ARS programming of PBX-1.
3. PBX-2 sends the call to the outside party "01-23-4567".

c) Blocking of Trunk Call through Other PBXs and How to Override It

To prohibit a TIE line call to go through PBX-2 for making a trunk call, if PBX-2 is Panasonic Hybrid IP-PBX (e.g., KX-TDA100 or KX-TDA200), PBX-2 should disable the trunk group of the outgoing call from PBX-2 against the COS of the trunk group of the incoming call to PBX-2 (→ Trunk Group Number [500]) like the programming example shown below. TRS/Barring applies to a TIE line call on a COS of the trunk group of the incoming call basis. To override this prohibition, access the DISA floating extension number of PBX-2 and enter a verified code to change the COS temporarily.

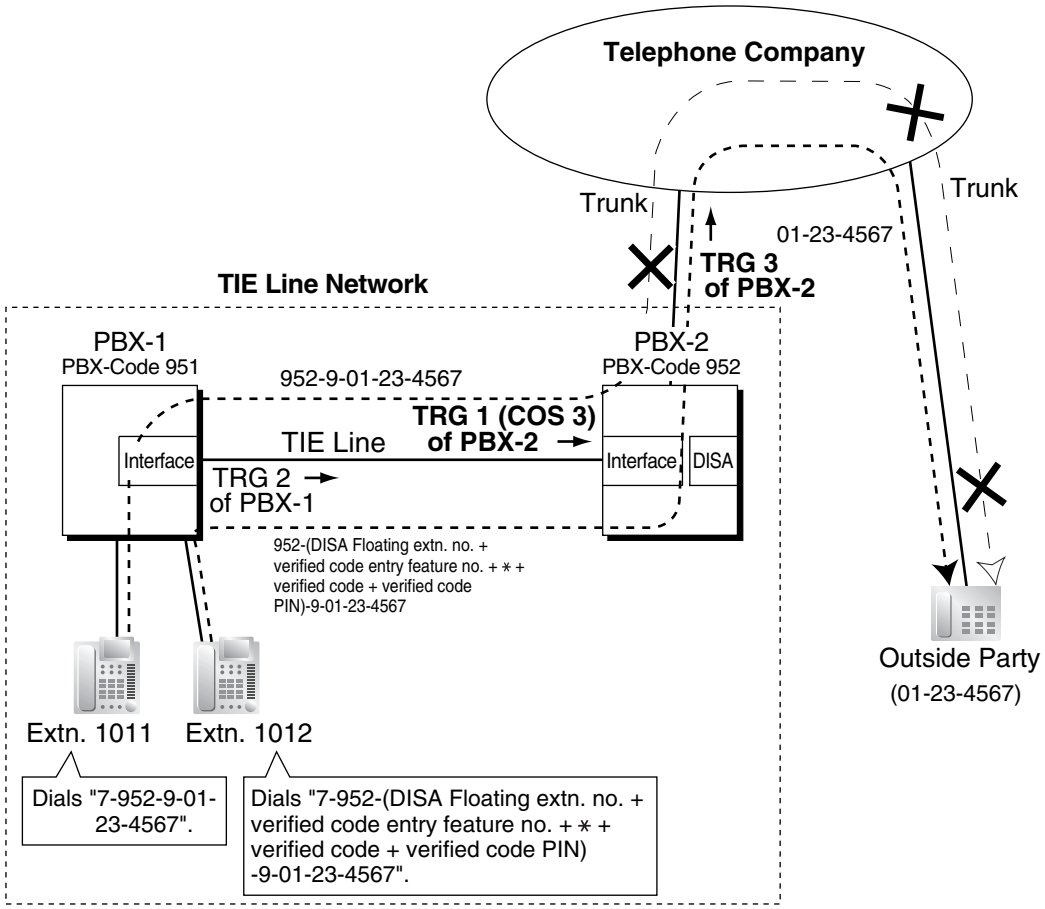
[Programming Example of PBX-2]

Trunk Group No.	COS No.
1	3
2	2
3	2
:	:

TRG of Incoming Call	Outgoing Call			
	TRG 1	TRG 2	TRG 3	…
COS 1		✔	✔	
COS 2	✔	✔		✔
COS 3		✔	✔	✔
:	:	:	:	:

✔ : Block

[Example]



Explanation:

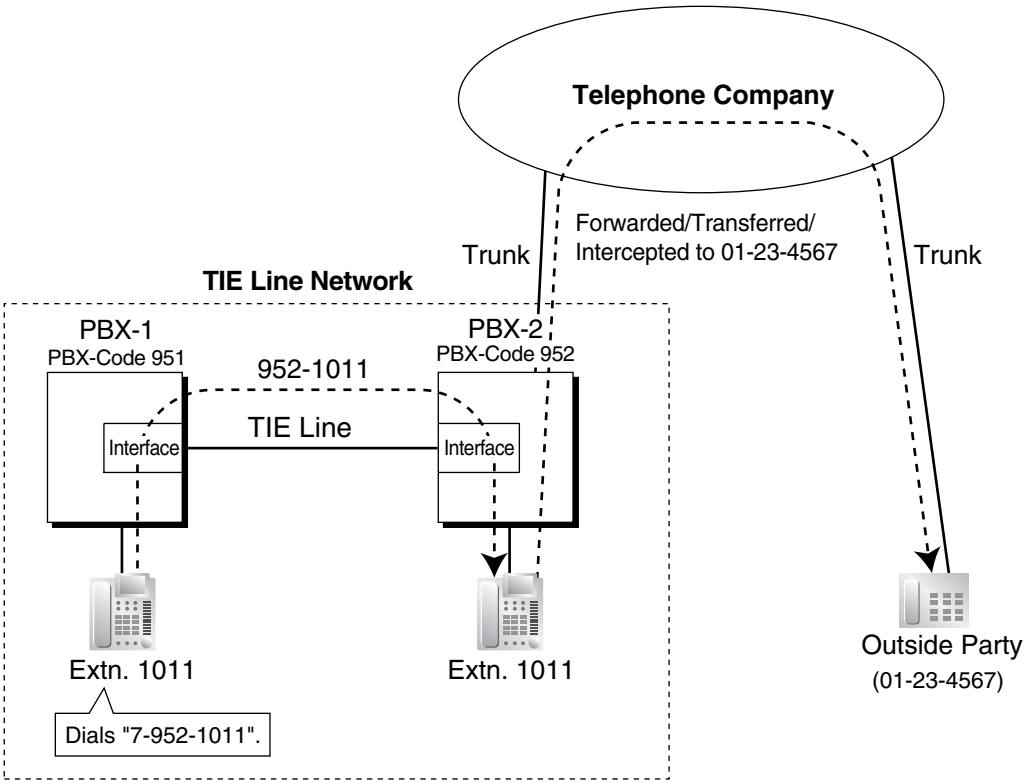
Case 1:

1. Extension 1011 of PBX-1 dials the TIE line access number "7", PBX code "952", Idle Line Access number of PBX-2 "9", and telephone number "01-23-4567".
2. The call is not connected to the outside party "01-23-4567" through PBX-2 which has PBX code "952" due to the COS assignment of the trunk groups of PBX-2.

Case 2:

1. Extension 1012 of PBX-1 dials the TIE line access number "7", PBX code "952", "DISA floating extension number of PBX-2 + verified code entry feature number + ✳ + verified code + verified code personal identification number (PIN)", Idle Line Access number of PBX-2 "9", and telephone number "01-23-4567".
2. The call overrides the COS assignment of the trunk groups of PBX-2, and the call is connected to the outside party "01-23-4567" through PBX-2 which has PBX code "952".

d) FWD/Call Transfer/Intercept Routing to the Trunk
[Example]



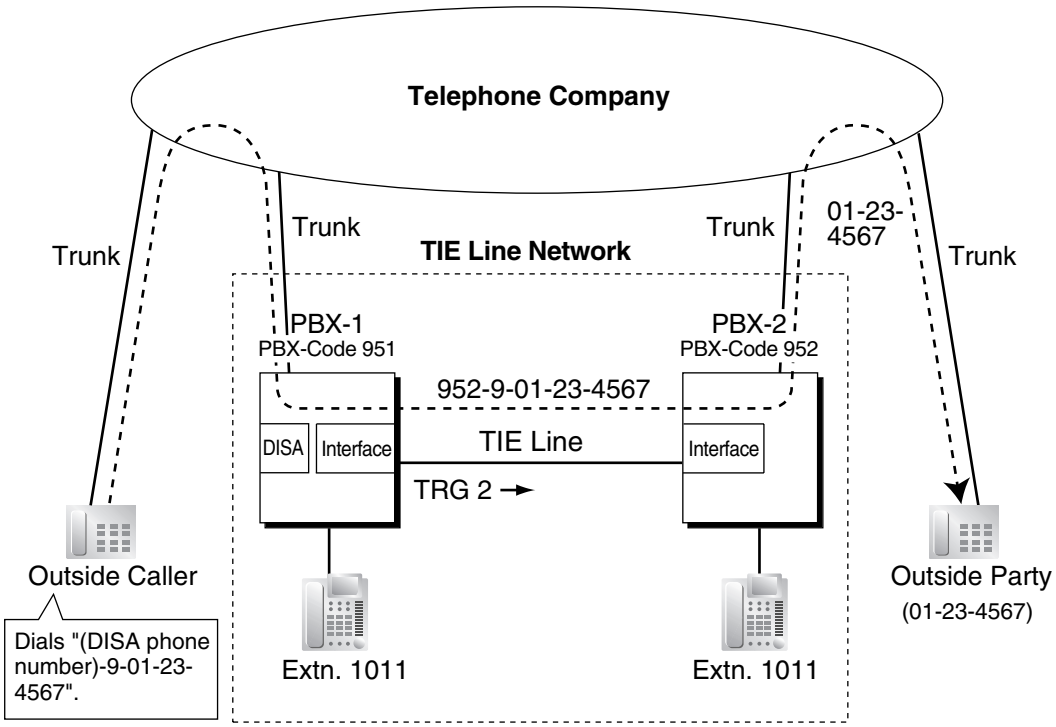
Explanation:

1. Extension 1011 of PBX-1 dials the TIE line access number "7", PBX code "952", and extension number "1011".
2. The call reaches the destination (extension 1011 of PBX-2) through the TIE line, and the call is forwarded, transferred or intercepted to the outside party "01-23-4567" through the trunk.

3.3 Trunk-to-TIE-to-Trunk Access

An outside caller can be connected to an outside party through the TIE line by using the DISA feature.

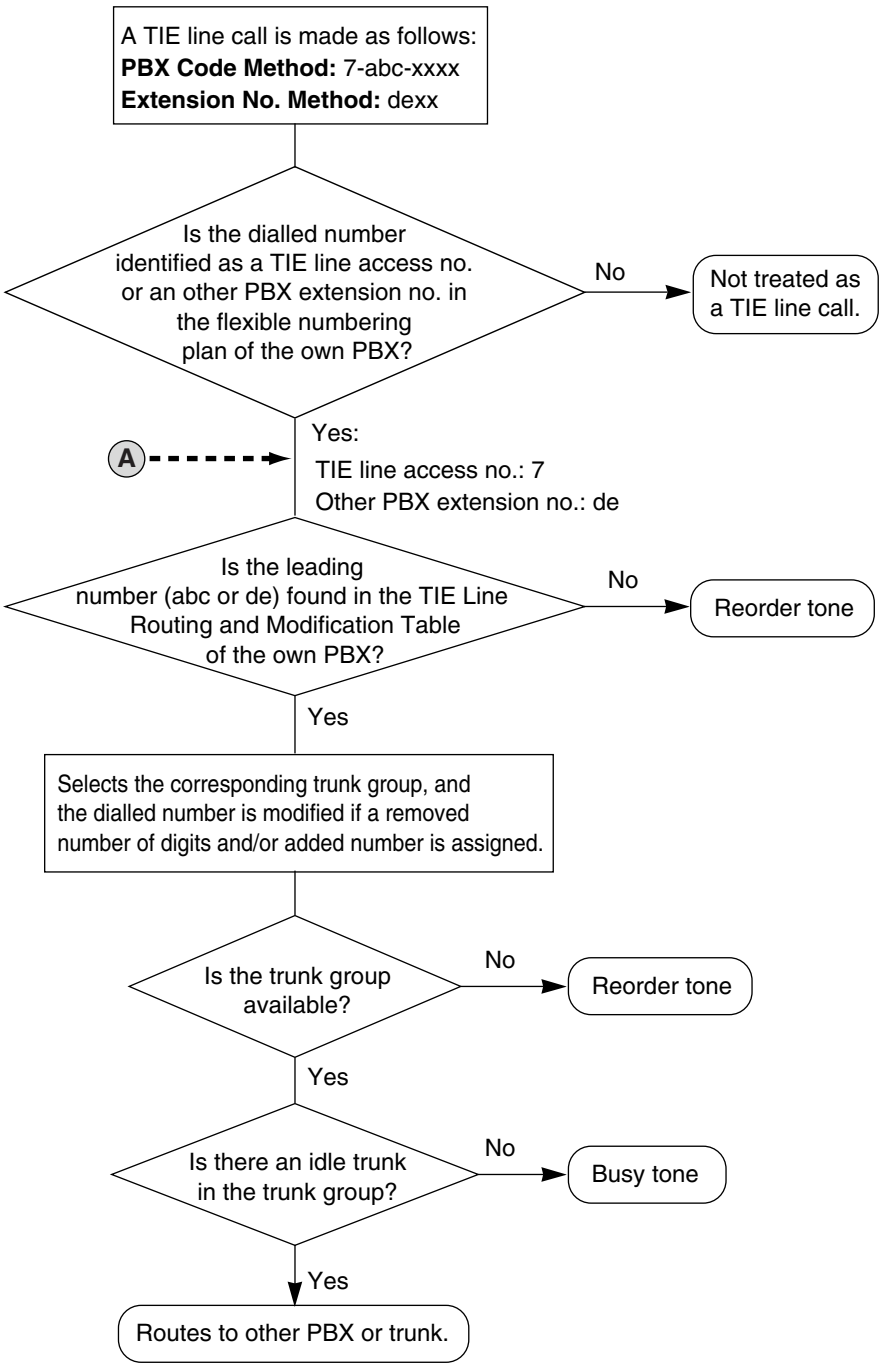
[Example]



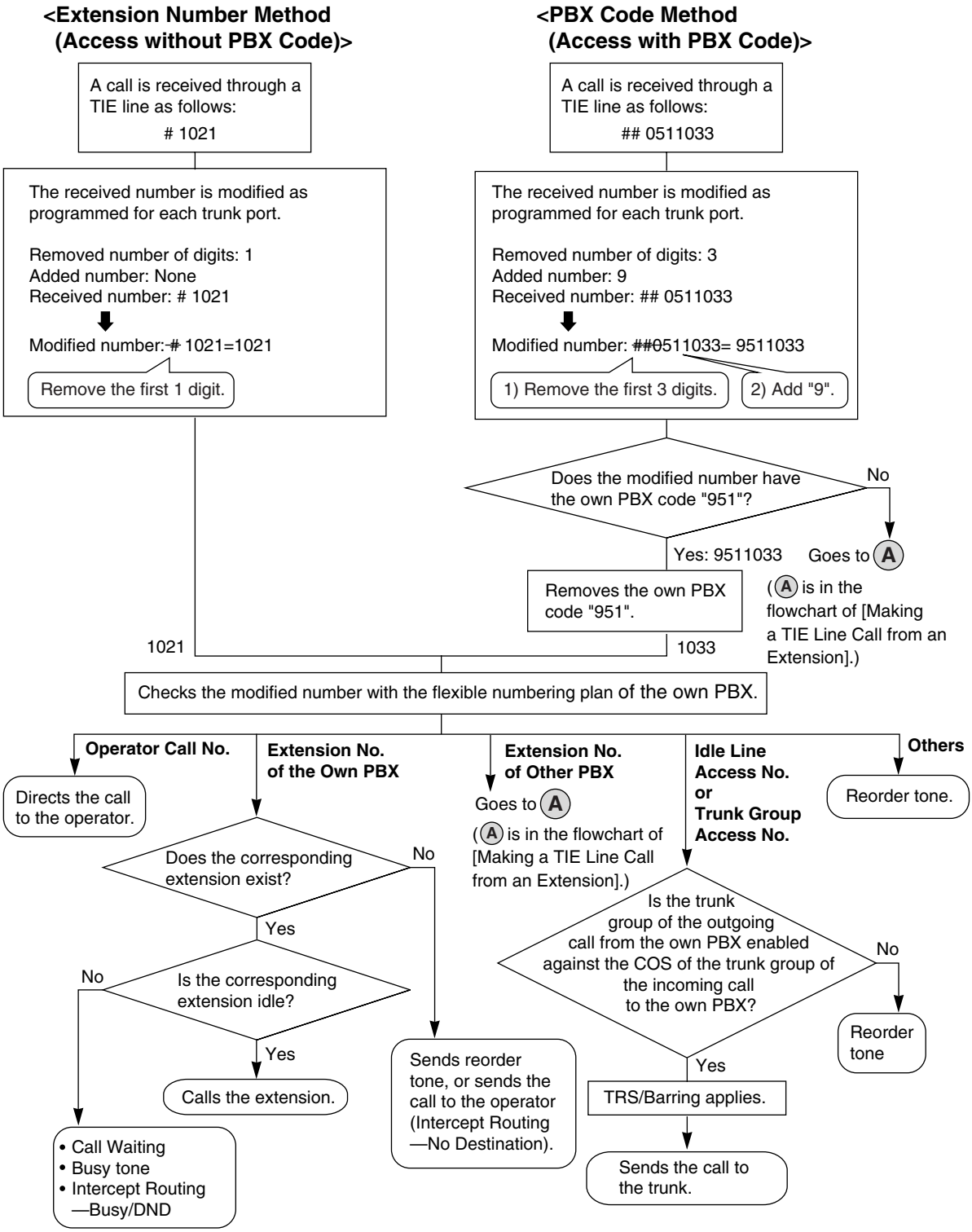
Explanation:

1. The outside caller dials the "DISA phone number of PBX-1", Idle Line Access number of PBX-1 "9", and telephone number "01-23-4567".
2. PBX-1 modifies the call (adds "952" and the Idle Line Access number of PBX-2 "9") and sends the call to PBX-2 which has PBX code "952" through the TIE line (trunk group [TRG] 2) according to the ARS programming of PBX-1.
3. PBX-2 sends the call to the outside party "01-23-4567".

4. TIE Line Routing Flowchart
[Making a TIE Line Call from an Extension]


A TIE line call is made as follows:
PBX Code Method: 7-abc-xxxx
Extension No. Method: dexx
Is the dialled number identified as a TIE line access no. or an other PBX extension no. in the flexible numbering plan of the own PBX?
No
Not treated as a TIE line call.
Yes:
TIE line access no.: 7
Other PBX extension no.: de
A
Is the leading number (abc or de) found in the TIE Line Routing and Modification Table of the own PBX?
No
Reorder tone
Yes
Selects the corresponding trunk group, and the dialled number is modified if a removed number of digits and/or added number is assigned.
Is the trunk group available?
No
Reorder tone
Yes
Is there an idle trunk in the trunk group?
No
Busy tone
Yes
Routes to other PBX or trunk.

[Receiving a Call through a TIE Line]


<Extension Number Method (Access without PBX Code)>
A call is received through a TIE line as follows:
1021
The received number is modified as programmed for each trunk port.
Removed number of digits: 1
Added number: None
Received number: # 1021
Modified number: # 1021=1021
Remove the first 1 digit.
<PBX Code Method (Access with PBX Code)>
A call is received through a TIE line as follows:
0511033
The received number is modified as programmed for each trunk port.
Removed number of digits: 3
Added number: 9
Received number: ## 0511033
Modified number: ##0511033= 9511033
1) Remove the first 3 digits.
2) Add "9".
Does the modified number have the own PBX code "951"?
No
Yes: 9511033
Goes to A
(A is in the flowchart of [Making a TIE Line Call from an Extension].)
Removes the own PBX code "951".
1021
1033
Checks the modified number with the flexible numbering plan of the own PBX.
Operator Call No.
Directs the call to the operator.
Extension No. of the Own PBX
Does the corresponding extension exist?
No
Yes
No
Is the corresponding extension idle?
Yes
Calls the extension.
• Call Waiting
• Busy tone
• Intercept Routing —Busy/DND
Extension No. of Other PBX
Goes to A
(A is in the flowchart of [Making a TIE Line Call from an Extension].)
Sends reorder tone, or sends the call to the operator (Intercept Routing —No Destination).
Idle Line Access No. or Trunk Group Access No.
Is the trunk group of the outgoing call from the own PBX enabled against the COS of the trunk group of the incoming call to the own PBX?
No
Reorder tone
Yes
TRS/Barring applies.
Sends the call to the trunk.
Others
Reorder tone.

Conditions

- When a TIE line call arrives at a busy extension which has disabled Call Waiting, the caller will hear busy tone. If required, Intercept Routing—Busy/DND can be activated.
- The Inter-digit time can be assigned for TIE line calls.

Feature Guide References

1.1.1.6 Intercept Routing
1.3.1.2 Call Forwarding (FWD)
1.8.1 Toll Restriction (TRS)/Call Barring (Barring)
1.9.1 Automatic Route Selection (ARS)
1.11.1 Call Transfer
1.15.6 Direct Inward System Access (DISA)
2.3.5 Flexible Numbering/Fixed Numbering

User Manual References

User Manual

1.2.1 Basic Calling

1.27.2 Virtual Private Network (VPN)

Description

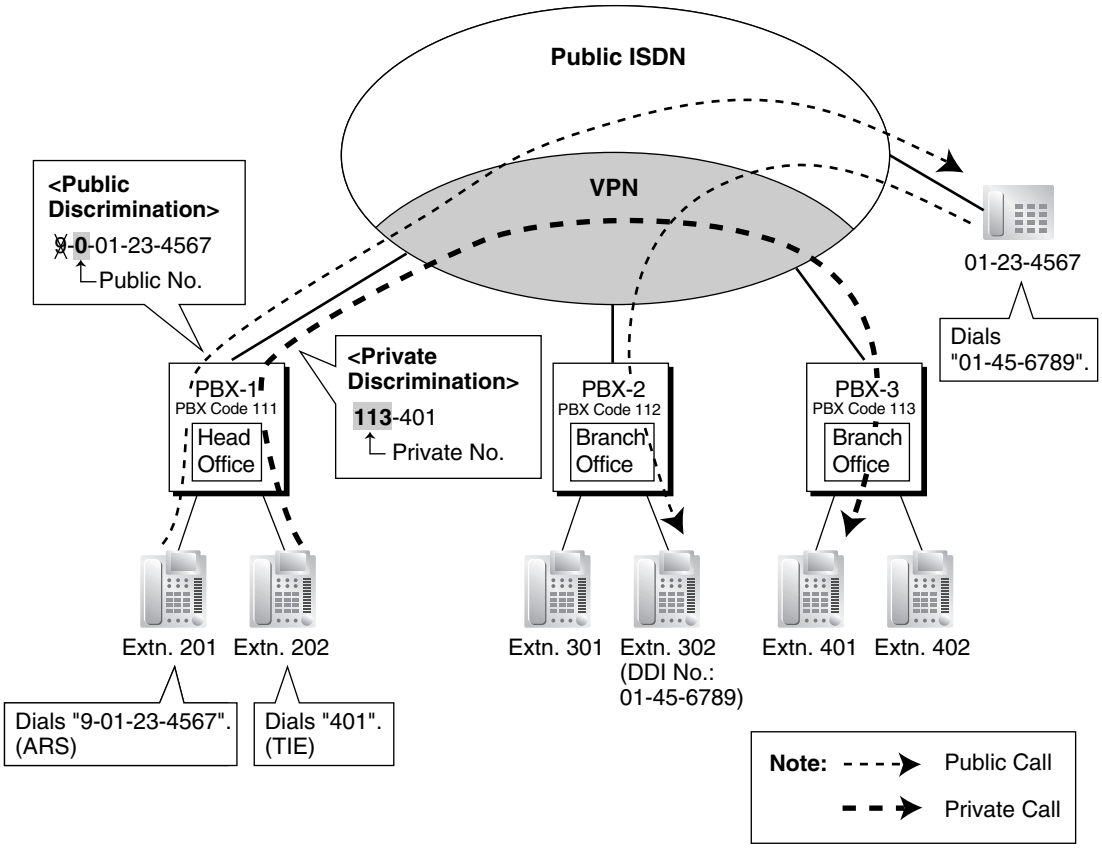
Virtual Private Network (VPN) is a service provided by the telephone company. It uses an existing line as if it were a private line. There is no need to set up a private line or to lease a line from the telephone company. Making and receiving both public and private calls is possible using the same line.

Public/Private Discrimination:

a) **When making a call:** The public/private discrimination number is required before sending the dialled number to the telephone company. The public/private discrimination number can be dialled manually, or automatically by ARS programming (→ 1.9.1 Automatic Route Selection (ARS)) and/or TIE line service programming.

b) **When receiving a call:** The telephone company distinguishes the call type. If it is a private call, the call is received by the TIE line service method. If it is a public call, the call is received by the Incoming Trunk Call Distribution method (DIL/DDI/MSN) which is assigned on the trunk.

[Example]



Conditions

- Each trunk has its service type: public, private, or VPN. Select VPN to use this service through system programming.

- Even if the telephone company does not support the VPN service, it is possible to use the same kind of service when making a call by TIE line service programming, and/or Quick Dialling programming (→ 1.6.1.6 Quick Dialling).

  [Quick Dialling Programming Example]

Location No.	Quick Dialling No.	Desired No.
Quick Dialling 01	2345 (extension no. of other PBX)	9-123-4321 (Public no. of extension 2345)
:	:	:

  Explanation:
  When an extension user dials "2345", he is connected to extension "2345" of other PBX whose public number is "123-4321".

Feature Guide References

1.27.1 TIE Line Service

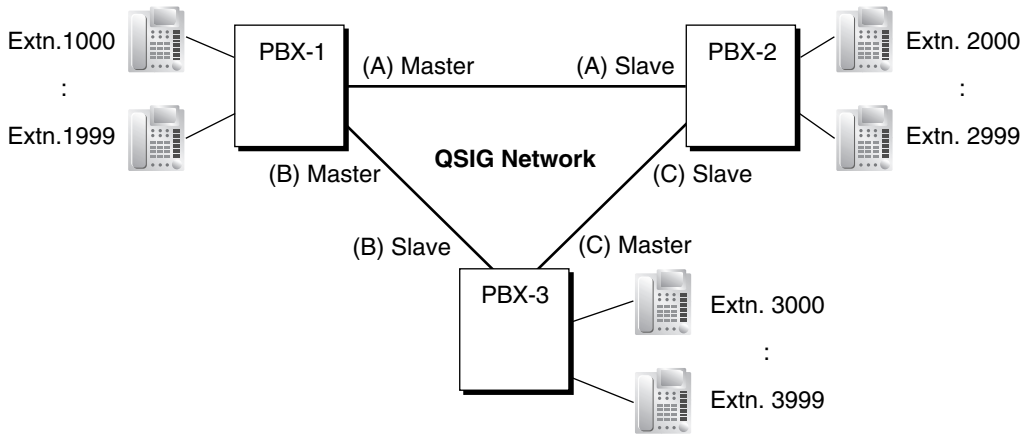
1.27.3 QSIG Network

1.27.3.1 QSIG Network—SUMMARY

Description

QSIG is a protocol which is based on ISDN (Q.931) and offers enhanced PBX features in a private network. The QSIG network supports private communications by the TIE line service method.
Each ISDN (QSIG) connection must have a master port on one Panasonic Hybrid IP-PBX and a slave port on the other Panasonic Hybrid IP-PBX to establish a QSIG network.

[Connection Example]



System programming controls the following services individually for each ISDN (QSIG) port.

[Service Table]

Service	Description	Details in
Calling Line Identification Presentation (CLIP)	Sends the caller's number to the QSIG network when making a call.	• 1.27.3.2 Calling/ Connected Line Identification Presentation (CLIP/COLP) and Calling/ Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Connected Line Identification Presentation (COLP)	Sends the number of the answered party to the QSIG network when answering a call.	
Calling Line Identification Restriction (CLIR)	Prevents the caller's CLI being presented to the called party by the caller.	
Connected Line Identification Restriction (COLR)	Prevents COLP being sent by the answered party.	

Service	Description	Details in
Calling Name Identification Presentation (CNIP)	Sends the caller's name to the QSIG network when making a call.	• 1.27.3.2 Calling/ Connected Line Identification Presentation (CLIP/COLP) and Calling/ Connected Name Identification Presentation (CNIP/CONP)—by QSIG
Connected Name Identification Presentation (CONP)	Sends the name of the answered party to the QSIG network when answering a call.	
Calling Name Identification Restriction (CNIR)	Prevents the caller's name being presented to the called party by the caller.	
Connected Name Identification Restriction (CONR)	Prevents CONP being sent by the answered party.	
Call Forwarding (CF)—by QSIG*	Forwards a call to the QSIG network.	• 1.27.3.3 Call Forwarding (CF)—by QSIG
Call Transfer (CT)—by QSIG*	Transfers a call to the QSIG network.	• 1.27.3.4 Call Transfer (CT)—by QSIG
Completion of Calls to Busy Subscriber (CCBS)—by QSIG*	Receives callback ringing when a busy called party on the QSIG network becomes free.	• 1.27.3.5 Completion of Calls to Busy Subscriber (CCBS)—by QSIG

*: This feature is not available for the PRI23 card.

Feature Guide References

1.19.1 Integrated Services Digital Network (ISDN)
1.27.1 TIE Line Service

1.27.3.2 Calling/Connected Line Identification Presentation (CLIP/COLP) and Calling/Connected Name Identification Presentation (CNIP/CONP)—by QSIG

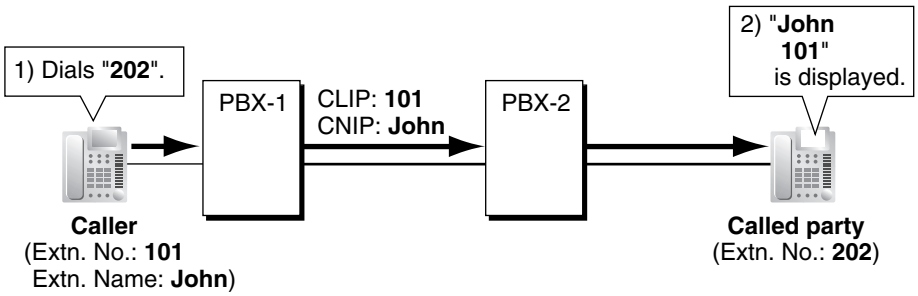
Description

Calling Line/Name Identification Presentation (CLIP/CNIP):
The PBX can send a preprogrammed extension number and/or name to the QSIG network when an extension user makes a call. The called party can see the number and/or name on his telephone display before answering the call.

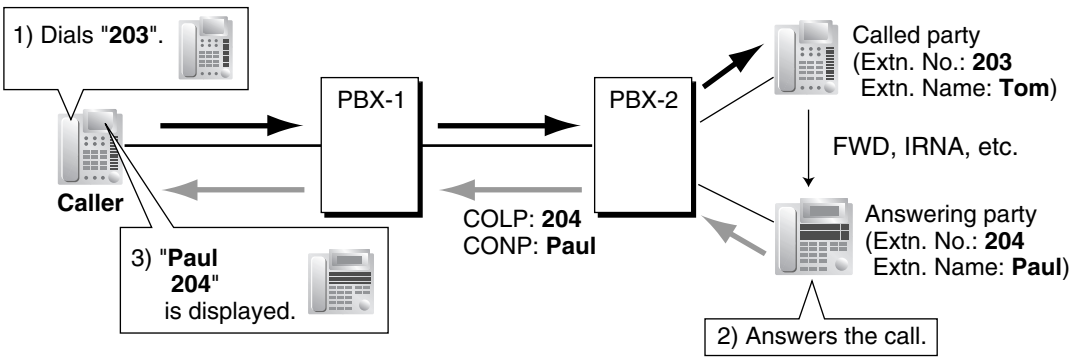
Connected Line/Name Identification Presentation (COLP/CONP):
The PBX sends a preprogrammed extension number and/or name to the QSIG network when the extension user answers an incoming call. The caller can see the number and/or name of the answering party on his telephone display when the call is answered.

These features comply with the following European Telecommunication Standard (ETS) specifications:
CLIP/COLP: ETS 300 172 Circuit mode basis services.
CNIP/CONP: ETS 300 238 Name identification supplementary services.

[CLIP/CNIP Example]



[COLP/CONP Example]



CLIP/COLP Number:
The extension number sent to the QSIG network for CLIP/COLP can be assigned for each extension through system programming (→ Extension Number [003]).

CNIP/CONP Name:
The extension name sent to the QSIG network for CNIP/CONP can be assigned for each extension through system programming (→ Extension Name [004]).

Calling/Connected Line Identification Restriction (CLIR/COLR):
It is possible for each extension to restrict the sending of its extension number to the QSIG network by pressing the CLIR button, COLR button, or entering the feature number.

Calling/Connected Name Identification Restriction (CNIR/CONR):
It is possible for each extension to restrict the sending of its extension name to the QSIG network. When CLIR is activated, CNIR becomes active automatically. When COLR is activated, CONR becomes active automatically.

These features comply with the following European Telecommunication Standard (ETS) specifications:
CLIR/COLR: ETS 300 172 Circuit mode basis services.
CNIR/CONR: ETS 300 238 Name identification supplementary services.

Conditions

- **COLP/CLIR/COLR/CNIP/CONP/CNIR/CONR Assignment for Each Port**
  Each service can be enabled or disabled on each ISDN (QSIG) port of the PBX.
- **CLIR Button and COLR Button**
  It is possible to switch between CLIP and CLIR by pressing the CLIR button, and COLP and COLR by pressing the COLR button. A flexible button can be customised as the CLIR or COLR button.

Feature Guide References

1.18.2 Flexible Buttons

User Manual References

User Manual

1.7.4 Displaying Your Telephone Number on the Called Party and Caller's Telephone (Calling/Connected Line Identification Presentation [CLIP/COLP])

1.7.5 Preventing Your Telephone Number Being Displayed on the Caller's Telephone (Connected Line Identification Restriction [COLR])

1.7.6 Preventing Your Number Being Displayed on the Called Party's Telephone (Calling Line Identification Restriction [CLIR])

1.27.3.3 Call Forwarding (CF)—by QSIG

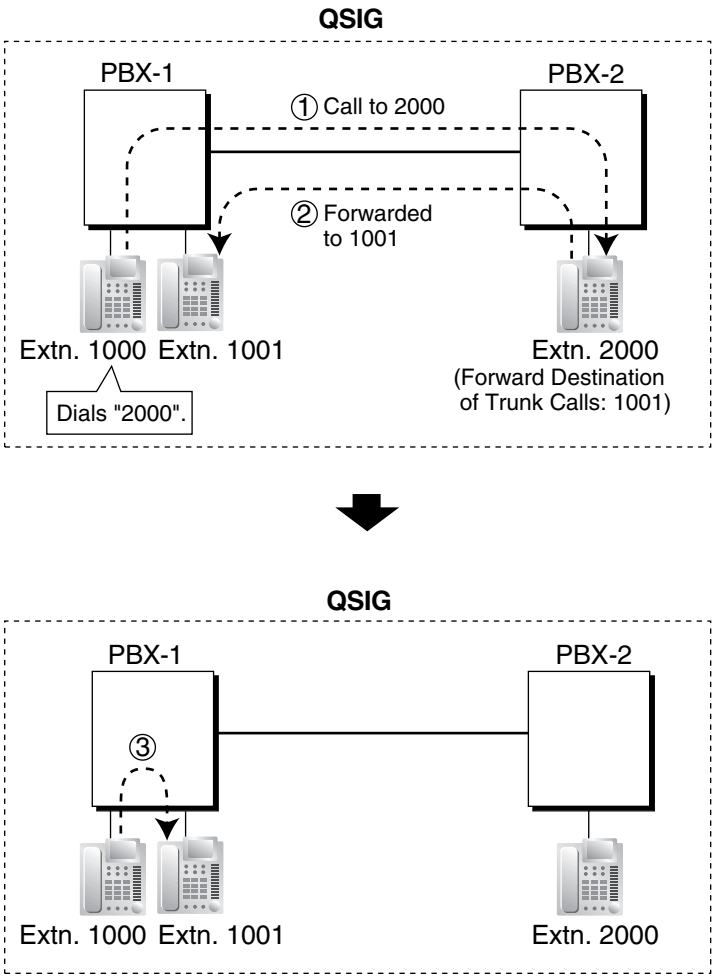
Description

The PBX forwards the call to a destination extension in another PBX in QSIG network. The destination can be set on your own PBX on an extension basis as the forward destination of trunk calls (→ 1.3.1.2 Call Forwarding (FWD)).

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 257 Diversion supplementary services.

If the same trunk group is used for the incoming call and the forwarded call, the following situation will be possible.

[Example]



① Extension 1000 of PBX-1 dials extension number "2000", and the call is sent to extension "2000" of PBX-2 through QSIG network.

② The call is forwarded to the forward destination of trunk calls of extension 2000, which is extension "1001" of PBX-1.

③ The call between PBX-1 and PBX-2 is released, and the call is connected directly to the forward destination of extension 2000.

Conditions

- This feature can be enabled or disabled on each ISDN (QSIG) port of the PBX.

User Manual References

User Manual

1.5.1 Forwarding Calls

1.27.3.4 Call Transfer (CT)—by QSIG

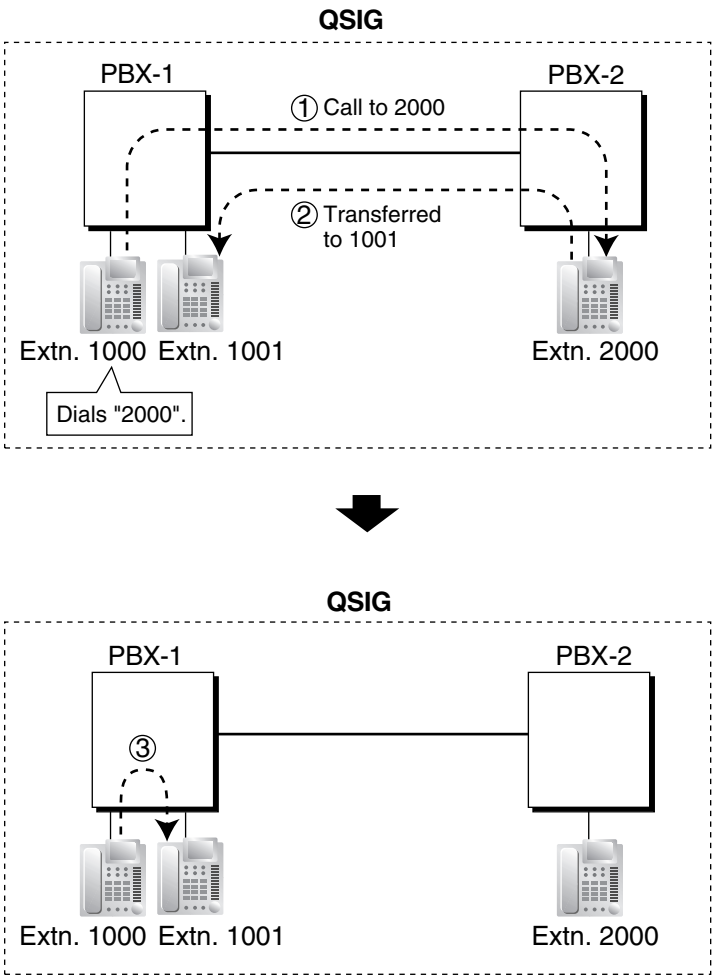
Description

The PBX transfers the call to a destination extension in another PBX in QSIG network.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 261 Call transfer supplementary service.

If the same trunk group is used for the incoming call and the transferred call, the following situation will be possible.

[Example]



① Extension 1000 of PBX-1 dials extension number "2000", and the call is sent to extension "2000" of PBX-2 through QSIG network.

② The call is transferred from extension 2000 to extension "1001" of PBX-1.

③ The call between PBX-1 and PBX-2 is released, and the call is connected directly to the transfer destination of extension 2000.

Conditions

- This feature can be enabled or disabled on an ISDN (QSIG) port basis.
- Call Transfer with Announcement and Call Transfer without Announcement is possible. (→ 1.11.1 Call Transfer)

User Manual References

User Manual

1.4.1 Transferring a Call (Call Transfer)

1.27.3.5 Completion of Calls to Busy Subscriber (CCBS)—by QSIG

Description

If the call has been made to an extension in another PBX in QSIG network and the called party is busy, an extension user can set to receive callback ringing when the called party becomes free. When the user answers the callback ringing, that party's number is automatically dialled.

This feature complies with the following European Telecommunication Standard (ETS) specifications: ETS 300 366 Call completion supplementary services.

Conditions

- This feature is available under the following conditions:
  - **a)** The caller's PBX is capable of using CCBS.
  - **b)** The called party's PBX is capable of accepting CCBS.
- To receive and send CCBS, receiving and sending CCBS must be enabled individually on an ISDN (QSIG) port basis through system programming.
- An extension user can set only one CCBS. The last setting is effective.
- The CCBS setting is cancelled if there is no callback ringing within 60 minutes or callback ringing is not answered within 10 seconds.

User Manual References

User Manual

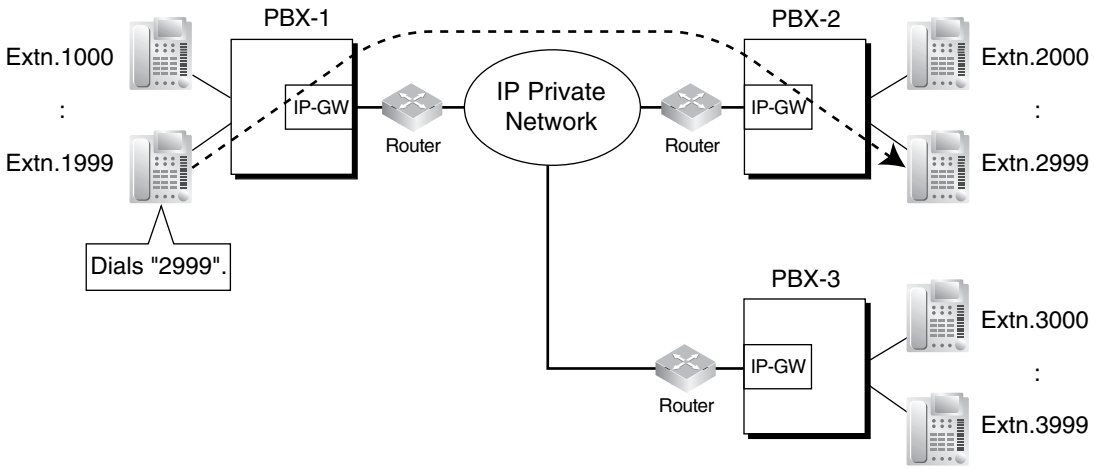
1.2.4 When the Dialled Line is Busy or There is No Answer

1.27.4 Voice over Internet Protocol (VoIP) Network

Description

The PBX can connect to another PBX via an IP-type private network. In this case, voice signals are converted into IP packets and sent through this network.
VoIP network supports the private network communications by the TIE line service method.

[Example]



[Required Programming]

Device	Programming
PBX	**For making a call:** • ARS programming (→ 1.9.1 Automatic Route Selection (ARS)) or TIE line service programming **For receiving a call:** • TIE line service programming
IP-GW (IP Gateway Circuit)	• IP address assignment for the own PBX and other PBXs.

Conditions

- QSIG service is available. (→ 1.27.3 QSIG Network)

Feature Guide References

1.27.1 TIE Line Service

1.28 Computer Telephony Integration (CTI) Features

1.28.1 Computer Telephony Integration (CTI)

Description

Connecting a personal computer (PC) to this PBX (via a DPT with the USB Module, or via a Server PC on a LAN) enables extension users to make use of advanced features by using the stored data in the PC or in the Server PC.

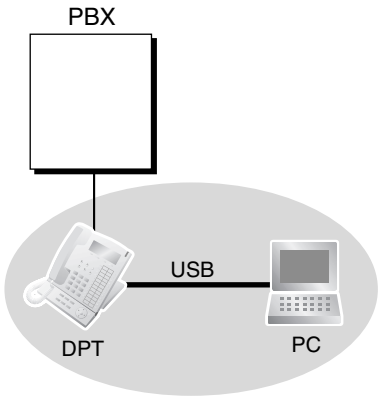
There are two types of CTI as follows:

1) First Party Call Control

2) Third Party Call Control

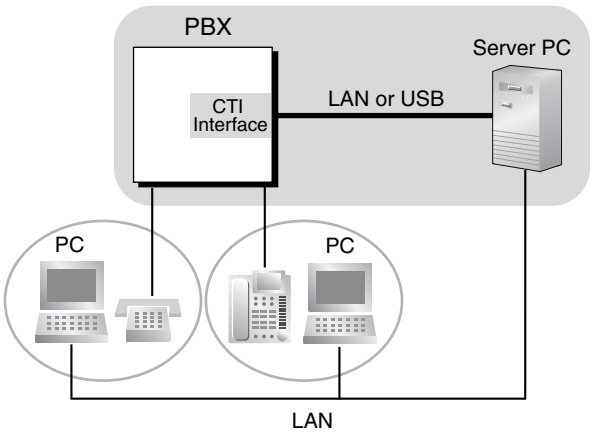
1. First Party Call Control

A PC is connected to a DPT using a USB port (USB Module) attached to the DPT. The PC monitors the status of the DPT and controls the DPT.



2. Third Party Call Control

A Server PC is connected to the PBX using the USB port on the PBX or using the CTI-LINK card as a CTI interface. PCs monitor the status of the PBX and control the PBX via the Server PC.



Conditions

- **Application Programming Interface (API)/Protocol**
  The following interface is required for First Party Call Control and Third Party Call Control:

Type	API/Protocol
First Party Call Control	• ECMA CSTA Phase 3
Third Party Call Control	• ECMA CSTA Phase 3 • TAPI 2.1

- The operating system of PC or Server PC required for First Party Call Control or Third Party Call Control depends on your CTI application software. For details, refer to the manual for your CTI application software.
- One Server PC can be connected per PBX.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.3.1 MPR Card

2.6.6 CTI-LINK Card

2.7.4 First Party Call Control CTI Connection

2.11.1 Connection of Peripherals

<KX-TDA30>

6.6.5 First Party Call Control CTI Connection

6.10.1 Connection of Peripherals

Section 2

System Configuration and Administration Features

2.1 System Configuration—Hardware

2.1.1 Extension Port Configuration

Description

There are three types of extension ports as follows:

a) **DPT Port:**
With the KX-TDA100/KX-TDA200, DPT, DSS Console, or Panasonic VPS (KX-TVS/TVP series [DPT (Digital) Integration]) can be connected.
With the KX-TDA30, DPT, DSS Console, KX-TVS/TVP series (DPT [Digital] Integration), or CS can be connected.

b) **SLT Port:** SLT or KX-TVS/TVP series (DTMF Integration) can be connected.

c) **Super Hybrid Port:**
With the KX-TDA100/KX-TDA200, DPT, APT, DSS Console, KX-TVS/TVP series, or SLT can be connected.
With the KX-TDA30, DPT, APT, DSS Console, KX-TVS/TVP series, SLT, or CS can be connected.

EXtra Device Port (XDP) of Super Hybrid Ports:
A DPT and SLT can be connected to one super hybrid port (TR: SLT, HL: DPT). In this case, the SLT port (TR) of the super hybrid port can be used as an XDP port to connect an SLT as a sub telephone. There are two modes for the XDP port as follows:

Mode	Description
Parallel Mode	The DPT and SLT have the **same extension number** so that they can act as one extension. They use the main telephone's (DPT's) extension data (e.g., extension number, COS). (→ 1.10.9 Parallelled Telephone)
XDP Mode	The DPT and SLT have **different extension numbers** so that they can act as completely different extensions. To use XDP mode, XDP mode must be enabled (on) the port through system programming (→ EXtra Device Port (XDP) Mode [600]).

Conditions

- **Automatic Detection on Super Hybrid Port**
  With the KX-TDA100/KX-TDA200, a DPT, APT, or SLT can be connected to a super hybrid port without programming.
  With the KX-TDA30, a DPT, APT, SLT, or CS can be connected to a super hybrid port without programming.
- A DSS Console or a VPS (Panasonic KX-TVS/TVP series [DPT (Digital) Integration]) can also be connected with an SLT in XDP mode.
- **APT and SLT in Parallel Mode**
  An APT and an SLT can also be connected to a super hybrid port and used in parallel mode.
- **Digital XDP (KX-TDA30 only)**
  A DPT can be connected to another DPT and act as a completely different extension. (→ 1.10.9 Parallelled Telephone)

- **Wireless XDP Parallel Mode**
  A PS can be used in parallel mode with a wired telephone.
  (→ 1.23.5 Wireless XDP Parallel Mode)
- **DSS Console and Paired Telephone Assignment**
  When a DSS Console is connected, a paired extension must be assigned through system programming (→ Console Paired Telephone [007]). Only a PT can be a paired extension.

2.2 System Configuration—Software

2.2.1 Class of Service (COS)

Description

Each extension is assigned a COS number (→ Class of Service [602]). The following features are controlled on a COS basis:

a) Internal Call Block (→ 1.1.2.2 Internal Call Block)
b) FWD (→ 1.3.1.2 Call Forwarding (FWD))
c) DND Override (→ 1.3.1.3 Do Not Disturb (DND))
d) Call Pickup (→ 1.4.1.3 Call Pickup)
e) Account Code Entry (→ 1.5.4.3 Account Code Entry)
f) External Call Block (→ 1.5.5.3 Trunk Access)
g) Executive Busy Override (→ 1.7.2 Executive Busy Override)
h) Call Monitor (→ 1.7.3 Call Monitor)
i) OHCA (→ 1.7.4.3 Off-hook Call Announcement (OHCA))
j) Whisper OHCA (→ 1.7.4.4 Whisper OHCA)
k) TRS/Call Barring (→ 1.8.1 Toll Restriction (TRS)/Call Barring (Barring))
l) Extension Lock (→ 1.8.3 Extension Lock)
m) Walking COS (→ 1.8.5 Walking COS)
n) Trunk Call Limitation (→ 1.10.8 Trunk Call Limitation)
o) Call Transfer (→ 1.11.1 Call Transfer)
p) Door Open (→ 1.15.2 Door Open)
q) DISA (→ 1.15.6 Direct Inward System Access (DISA))
r) Call Forwarding (CF)—by ISDN (P-MP) (→ 1.19.1.4 Call Forwarding (CF)—by ISDN (P-MP))
s) Wireless XDP Parallel Mode (→ 1.23.5 Wireless XDP Parallel Mode)
t) SMDR for Outgoing Trunk Call (→ 1.24.1 Station Message Detail Recording (SMDR))
u) Time Service Switching (→ 2.2.4 Time Service)
v) Manager Features (→ 2.2.6 Manager Features)
w) PT Programming (→ 2.3.2 PT Programming)

Conditions

- **Walking COS**
  Extension users can make a call from other extensions of the lower level COS by using their own higher level COS temporarily.

User Manual References

User Manual

1.2.7 Calling without Restrictions

2.2.2 Group

Description

This PBX supports various types of groups.

1. Trunk Group

Trunks can be grouped into a specified number of trunk groups (e.g., for each carrier, trunk type, etc.) (→ LCOT/BRI Trunk Group Number [402]). Several settings can be assigned on a trunk group basis. All trunks belonging to a trunk group follow the assignment determined for that trunk group.
One trunk can belong to only one trunk group on a port or channel basis.
Port basis: LCOT/DID/E & M/ISDN-BRI/ISDN-PRI23/ISDN-PRI30
Channel basis: E1/T1

2. User Group

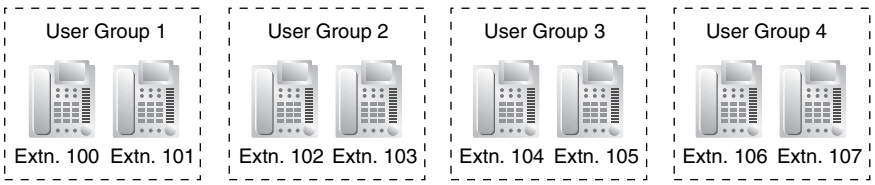
The PBX supports user groups (→ User Group [603]), each of which is used to compose the following groups:

a) Tenant (→ 2.2.3 Tenant Service)

b) Call Pickup Group (See below.)

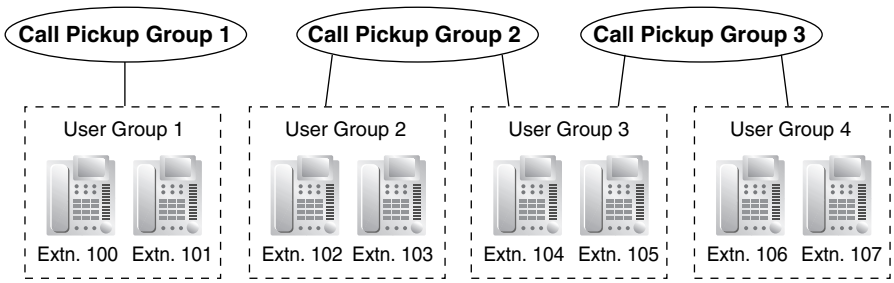
c) Paging Group (See below.)

Every extension must belong to one user group, but cannot belong to more than one user group.
Assignable Extensions: PT/SLT/PS/ISDN Extension/T1-OPX
[Example]



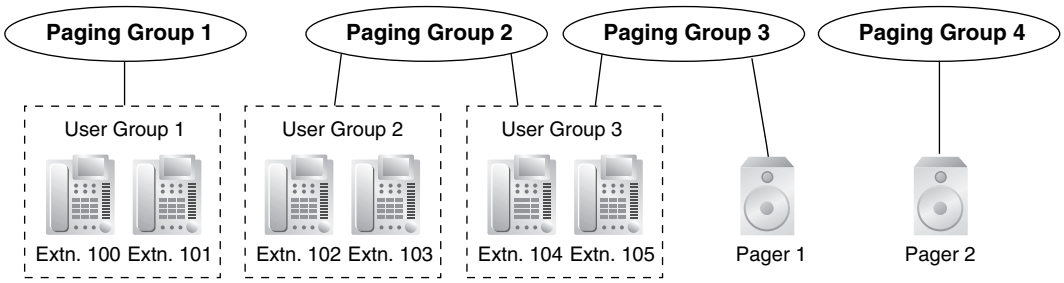
2.1 Call Pickup Group (→ User Groups of a Pickup Group [650])

Using the Group Call Pickup feature, extensions can answer any calls within a specified group. One user group can belong to several call pickup groups. (→ 1.4.1.3 Call Pickup)
[Example]



2.2 Paging Group (→ User Groups of a Paging Group [640])

Using the Paging feature, extensions can page or answer a page within their groups. One user group or external pager can belong to several paging groups.
(→ 1.14.1 Paging)
[Example]

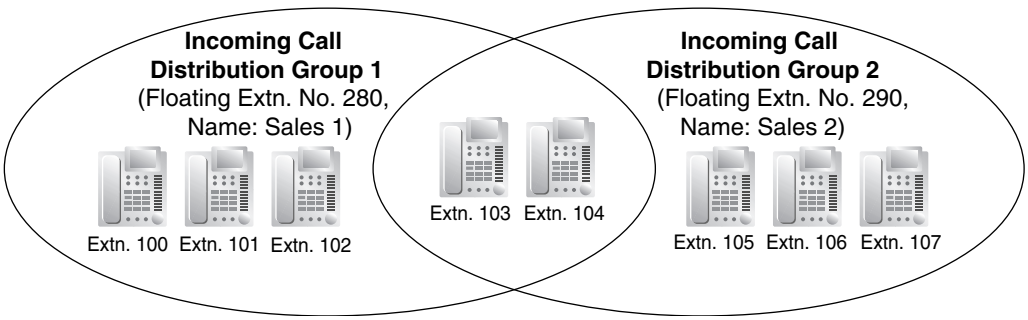


3. Idle Extension Hunting Group

If a called extension is busy or in DND mode, Idle Extension Hunting redirects the incoming call to an idle member of the same idle extension hunting group, which can be programmed through system programming (→ Idle Extension Hunting Group Member [681]). Idle extensions are automatically searched according to a preprogrammed hunting type: Circular Hunting or Terminated Hunting (→ Idle Extension Hunting Type [680]).
(→ 1.2.1 Idle Extension Hunting)

4. Incoming Call Distribution Group

An incoming call distribution group is a group of extensions which receives incoming calls directed to the group (→ Incoming Call Distribution Group Member [620]). Each incoming call distribution group has a floating extension number (→ Incoming Call Distribution Group Floating Extension Number [622]) and name (→ Incoming Call Distribution Group Name [623]). One extension can belong to multiple groups.
Assignable Extensions: PT/SLT/PS/ISDN Extension/T1-OPX/PS Ring Group
(→ 1.2.2 Incoming Call Distribution Group Features)
[Example]


Incoming Call Distribution Group 1
(Floating Extn. No. 280, Name: Sales 1)
Incoming Call Distribution Group 2
(Floating Extn. No. 290, Name: Sales 2)
Extn. 100 Extn. 101 Extn. 102
Extn. 103 Extn. 104
Extn. 105 Extn. 106 Extn. 107

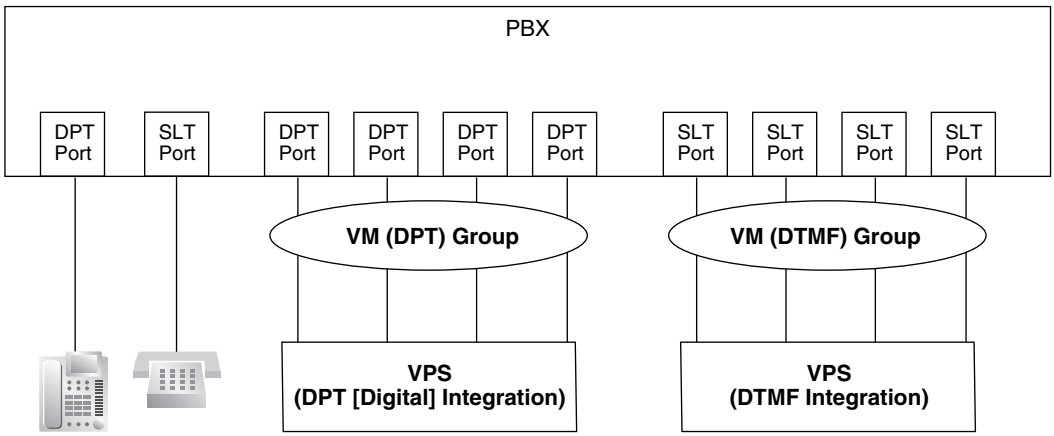
5. VM Group

There are two types of VM groups as follows:

Type	Description
VM (DTMF) Group	A group of SLT ports which use the Voice Mail DTMF Integration features. One SLT port can belong to only one group.
VM (DPT) Group	A group of DPT ports (→ VM Group Floating Extension Number [660]) which use the Voice Mail DPT (Digital) Integration features. One DPT port can belong to only one group.

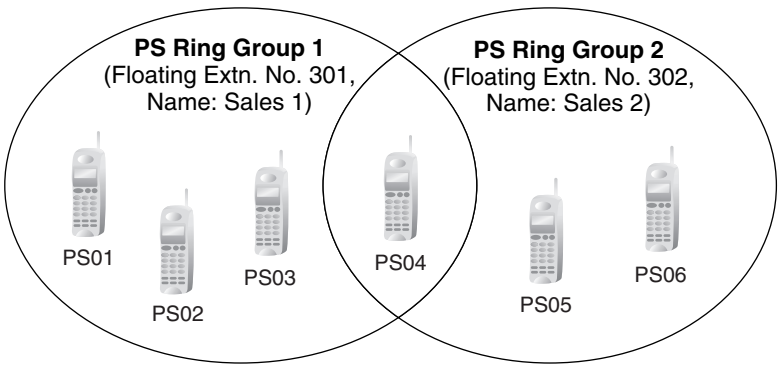
(→ 1.22.1 Voice Mail (VM) Group)

[Example]



6. PS Ring Group

PS ring group is a group of PS extensions which receives incoming calls directed to the group. Each group has a floating extension number and name through system programming. One PS can belong to multiple groups.
(→ 1.23.2 PS Ring Group)



2.2.3 Tenant Service

Description

This PBX can be shared with certain number of tenants.

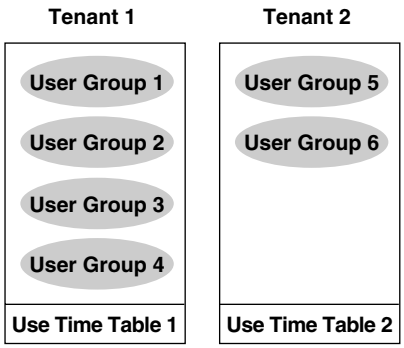
1. Tenant Configuration

1.1 Tenant Member

The tenant members consist of user groups. One user group can belong to only one tenant. Therefore, one extension can belong to only one tenant.
(→ 2.2.2 Group)

1.2 Time Service

Each tenant has a Time Table. The Start and/or End time of each time mode (day/lunch/break/night) can be set for each day of the week. The Time Table numbers correspond to the tenant numbers respectively.
(→ 2.2.4 Time Service)
[Example]



2. System Management

Each of the following system management items can be assigned to each tenant.

a) Tenant Operator (extension number/floating extension number of incoming call distribution group/none) (→ 2.2.5 Operator Features)

b) ARS Mode (Off/Local Access/All Access/System) (→ 1.9.1 Automatic Route Selection (ARS))

c) Music Source for Music on Hold (System/BGM Number/Cyclic Tone)
(→ 1.12.4 Music on Hold)

[Programming Example]

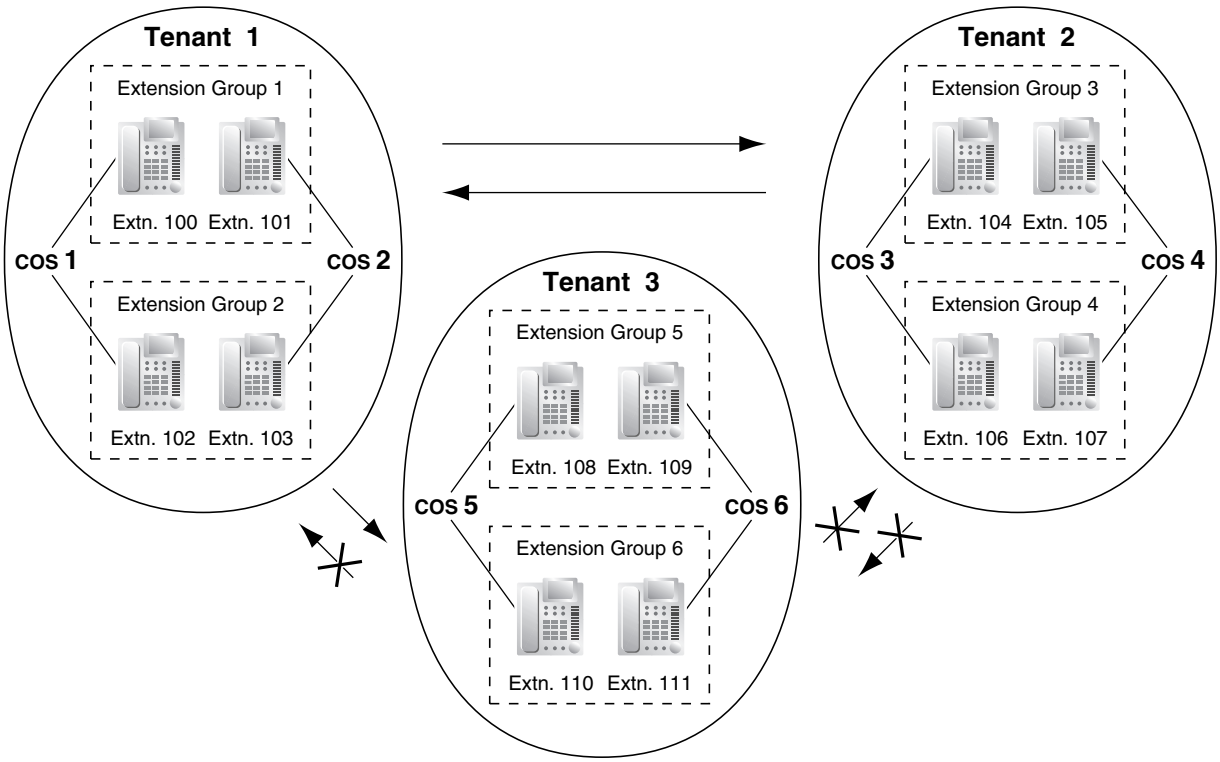
Tenant No.	Operator	ARS Mode	Music Source
1	Extn.101	Local Access	System*3
2	None*1	System*2	Cyclic Tone
3	Floating extn. no. 200	Off	BGM1
:	:	:	:

*1: Follows the system assignment of a PBX operator (→ Operator Assignment [006])
*2: Follows the system assignment of the ARS mode (→ ARS Mode [320])
*3: Follows the system assignment of the music source for the Music on Hold (→ Music on Hold [711])

Conditions

- **Tenant-to-Tenant Call Block**
  The following features can be restricted based on the COS for each extension (not based on the tenant) by the Internal Call Block feature (→ 1.1.2.2 Internal Call Block):
  - Calling extensions or doorphone(s) in the restricted tenant(s)
  - Picking up calls ringing in the restricted tenant(s)
  - Retrieving a call held within the restricted tenant(s).

[Example]



[Programming Example]

Caller	Called Party						
	COS 1	**COS 2**	**COS 3**	**COS 4**	**COS 5**	**COS 6**	**...**
COS 1							...
COS 2							...
COS 3					✔	✔	...
COS 4					✔	✔	...
COS 5	✔	✔	✔	✔			...
COS 6	✔	✔	✔	✔			...
:	:	:	:	:	:	:	:

✔: Block

Explanation:

1. Assign each extension in a tenant to a certain COS number. Each tenant must have unique COS numbers.

Tenant 1: COS 1 and COS 2

Tenant 2: COS 3 and COS 4

Tenant 3: COS 5 and COS 6

2. Tenant-to-Tenant Call Block enables by the Internal Call Block feature.

a) Tenant 1 (COS 1 and COS 2) can make calls to both Tenant 2 (COS 3 and COS 4) and Tenant 3 (COS 5 and COS 6) as well as Tenant 1.

b) Tenant 2 (COS 3 and COS 4) can make calls to Tenant 1 (COS 1 and COS 2) and Tenant 2.

c) Tenant 3 (COS 5 and COS 6) can make calls to Tenant 3 itself only.

- An incoming call distribution group must belong to one tenant because the following features are determined on a tenant basis (→ 1.2.2.1 Incoming Call Distribution Group Features—SUMMARY):
  - Music on Hold while a call is waiting in the queue
  - The Time Table which determines the overflow destination.

2.2.4 Time Service

Description

This PBX supports day, night, lunch, and break modes of operation. TRS/Barring can be arranged separately. The destination of incoming calls can be set differently for each mode.

1. Time Service Switching Mode

Day/lunch/break/night mode can be switched either automatically or manually (→ Time Service Switching Mode [101]). The switching mode can be assigned for each tenant. The switching mode can also be changed by pressing the Time Service Switching Mode (Automatic/Manual) button. This can be performed by only an extension assigned as the manager, or preprogrammed extension on a COS basis (→ Time Service Manual Switching [514]).

Type	Description
Automatic	The PBX will switch mode according to the preprogrammed Time Table.
Manual	A manager, or preprogrammed extension on a COS basis (→ Time Service Manual Switching [514]) can switch mode by dialling the feature number or pressing the Time Service button.

Even while in the Automatic Switching mode, day/lunch/break/night mode can be changed manually.

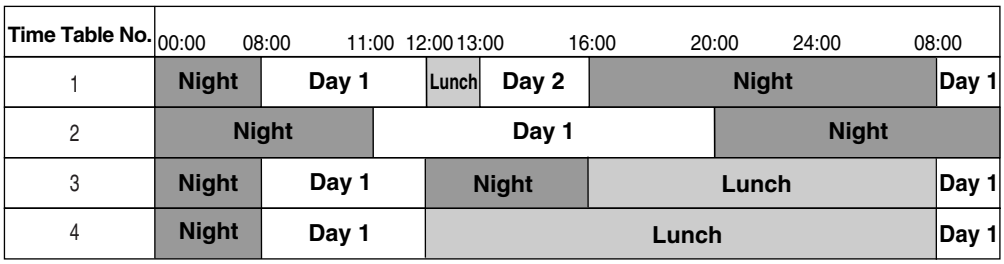
2. Time Table

Each tenant has a Time Table used for the Automatic Switching mode. The Start and/or End time of each mode can be set for each day of the week. The Time Table numbers correspond to the tenant numbers respectively.

[Time Table Example]

Time Schedule		Time Table No. (Tenant No.)				
		1	2	3	4	…
MON	Day 1 start	08:00	11:00	08:00	08:00	…
	Lunch start	12:00	NONE	16:00	12:00	…
	Day 2 start	13:00	NONE	NONE	NONE	…
	Break 1 start	NONE	NONE	NONE	NONE	…
	Break 1 end (Day restart)	NONE	NONE	NONE	NONE	…
	Night start	16:00	20:00	12:00	NONE	…
TUE	Day 1 start	08:00	11:00	08:00	08:00	…
	Lunch start	12:05	NONE	13:00	13:00	…
	Day 2 start	13:00	NONE	NONE	NONE	…
	Break 1 start	NONE	NONE	NONE	NONE	…
	Break 1 end (Day restart)	NONE	NONE	NONE	NONE	…
	Night start	16:31	20:00	17:00	NONE	…
:	:	:	:	:	:	…

<Time Service Image of Monday and Tuesday>



3. Features Using Time Service

The following features can be set in each time mode (day/lunch/break/night):

- **a)** Destination of incoming trunk calls (DIL/DID/DDI/MSN) (→ 1.1.1 Incoming Trunk Call Features)
- **b)** Destination of the Intercept Routing (→ 1.1.1.6 Intercept Routing)
- **c)** Queuing Time Table for incoming call distribution groups (→ 1.2.2.3 Queuing Feature)
- **d)** Overflow destination for incoming call distribution groups (→ 1.2.2.5 Overflow Feature)
- **e)** Destination of incoming doorphone calls (→ 1.15.1 Doorphone Call)
- **f)** PBX operator (→ 2.2.5 Operator Features)
- **g)** COS for TRS/Barring and for Trunk Access
- **h)** Outgoing Message (OGM) for Timed Reminder (→ 1.25.4 Timed Reminder)

[Programming Examples of DID/DDI Table and DIL Table]
DID/DDI table can be programmed for each DID/DDI number, and a tenant (Time Table) number is assigned to each DID/DDI number. DIL table can be programmed for each trunk, and a tenant (Time Table) number is assigned to each trunk.

<DID/DDI Table>

Location	DID/DDI No.	Tenant (Time Table) No.	DID/DDI Destination			
			Day	Lunch	Break	Night
001	123-4567	1	105	100 (VPS)	105	100 (VPS)
002	123-2468	1	102	100 (VPS)	102	100 (VPS)
:	:	:	:	:	:	:

<DIL Table>

Trunk No.	Tenant (Time Table) No.	DIL Destination			
		Day	Lunch	Break	Night
01	1	101	100 (VPS)	101	100 (VPS)
02	2	102	100 (VPS)	102	100 (VPS)
:	:	:	:	:	:

Explanation:
If a trunk call with a DID number (123-4567) is received at 20:00;

1) Tenant (Time Table) number 1 will be used.

2) The call is received during night mode in Time Table 1.

3) The call will be routed to the extension 100 (VPS).

4. Holiday Mode

The holiday mode activates automatically using the Automatic Switching mode. Up to 24 holidays (start and end dates) can be stored, and one time mode can be selected for all holidays.

5. Time Service Button

A flexible button can be customised as the following buttons:

a) Day/Night button

b) Day/Night/Lunch button

c) Day/Night/Break button

d) Day/Night/Lunch/Break button

Each of these buttons is used for switching between modes. For example, pressing the Day/Night button switches between day and night modes. All of these buttons show the current status as follows:

Light Pattern	Status
Off	Day mode
Red on	Night mode
Green on	Lunch mode
Slow green flashing	Break mode
Slow red flashing	Holiday mode

Note

Any extension user (except extension users allowed to change the mode) can only check the current status on the display by pressing the Time Service button.

Conditions

- PT programming (→ Time Service Starting Time [102]) can set the Start and/or End time of the following:
  - Day-1 (Day Start time)
  - Lunch (Lunch Start time)
  - Day-2 (Lunch End time)
  - Night (Night Start time)

  PC programming can also set the following three time periods for break mode per day.
  - Break-1 Start
  - Break-1 End (Day restart)
  - Break-2 Start
  - Break-2 End (Day restart)
  - Break-3 Start
  - Break-3 End (Day restart)

- **Time Service Switching Mode (Automatic/Manual) Button**
  A flexible button can be customised as the Time Service Switching Mode (Automatic/Manual) button.

Feature Guide References

1.8.1 Toll Restriction (TRS)/Call Barring (Barring)

1.18.2 Flexible Buttons

2.2.3 Tenant Service

User Manual References

User Manual

1.7.10 Checking the Time Service Status

2.1.2 Time Service Mode Control

2.2.5 Operator Features

Description

This PBX supports a PBX operator and a tenant operator. Any extension and incoming call distribution group can be designated as a PBX and/or a tenant operator.

Type	Description
PBX operator	An extension or incoming call distribution group can be assigned as a PBX operator for each time mode (day/lunch/break/night) (→ Operator Assignment [006]).
Tenant operator	An extension or incoming call distribution group can be assigned as a tenant operator. The tenant operator may be the extension or incoming call distribution group of another tenant. **[Example]** Extension 110 in tenant 1 is the tenant operator of tenant 3.

Operator Call:
An extension user can call an operator with a simple operation. When an operator call is made, the call is routed to the tenant operator. If no tenant operator exists, the call will be routed to the PBX operator. The time mode depends on the tenant of the calling extension. If neither the tenant operator nor the PBX operator exists, the caller will hear reorder tone.

Conditions

- An extension or incoming call distribution group can be assigned as both the tenant and PBX operator.
- A tenant operator can be assigned for multiple tenants.

Feature Guide References

2.2.3 Tenant Service

User Manual References

User Manual

1.2.1 Basic Calling

2.2.6 Manager Features

Description

An extension assigned as the manager (manager extension) is allowed to use the specified features. COS programming determines the extensions which can use the following manager features (→ Manager Assignment [511]):

Feature		Description	Manager Password	Details in
Manager Programming	**Manager Password Change**	Changes the manager password.	Required	–
	Call Charge Management	Sets, sees, clears, and prints the call charge data.	Required	• 1.24.2 Charge Meter
	Verified Code Personal Identification Number (PIN) Set	Sets a verified code PIN for each verified code.	Required	• 1.8.6 Verified Code Entry
	Remote PIN Clear	Clears the extension PIN of an extension remotely, and a verified code PIN. PIN Lock is also unlocked.	Required	• 1.25.1 Extension Personal Identification Number (PIN) • 1.8.6 Verified Code Entry
	Remote Extension Lock	Sets or cancels the Extension Lock on an extension remotely.	Required	• 1.8.3 Extension Lock
Dial Tone Transfer		Changes the TRS/Barring level of the extension temporarily. **[Example]** An extension user can call a manager to release the restricted outgoing call (e.g., international call).	Not required	• 1.8.4 Dial Tone Transfer
Outgoing Message (OGM)		Records and plays back outgoing messages (OGMs).	Not required	• 1.15.5 Outgoing Message (OGM)
Time Service		Switches the time mode (day/lunch/break/night) manually.	Not required	• 2.2.4 Time Service
BGM—External		Sets the External BGM on and off.	Not required	• 1.15.4 Background Music (BGM)

Conditions

WARNING
When a third party discovers the personal identification number (PIN) (verified code PIN/ extension PIN) of the PBX, there will be a risk that they will make fraudulent phone calls using the telephone line, and the cost will be charged to your own account. In order to avoid this problem, we strongly recommend the following points:

a) Carefully maintain the secrecy of the PIN.

b) Specify a complicated PIN as long and random as you can make it.

c) Change the PIN frequently.

- **Manager Password**
  One manager password can be assigned per PBX (→ Manager Password [112]).

User Manual References

User Manual

2.1 Control Features

3.2.2 Manager Programming

2.3 System Data Control

2.3.1 PC Programming

Description

This PBX can be programmed and administered using a personal computer (PC). There are two programming methods:

1) **On-site Programming:** System programming/diagnosis can be performed locally by connecting a PC to the PBX directly.
2) **Remote Programming:** System programming/diagnosis and data upload can be performed from a remote location.

1. On-site Programming:

Method	Description
Using the Serial Interface (RS-232C) port	The PBX has a Serial Interface (RS-232C) port which can be used for either system administration or SMDR (→ 1.24.1 Station Message Detail Recording (SMDR)).
Using the USB port	The PC is connected to the USB port on the PBX, or a USB port (USB Module) attached to a DPT.
Using the LAN interface (KX-TDA100/KX-TDA200 only)	A CTI-LINK card must be installed.
Using a modem through an SLT port*	An RMT card must be installed. Assign the floating extension number of the analogue remote maintenance (→ Modem Floating Extension Number [811]), and dial this number from the PC to connect to the PBX.
Using an ISDN TA interface (64 kbps) through an ISDN Extension Line*	Assign the floating extension number of the ISDN remote maintenance (→ ISDN Remote Floating Extension Number [812]), and dial this number from the PC to connect to the PBX. The RMT card is not required for this method. This method is available only when an user-supplied ISDN TA that supports CAPI is used.

*: If remote access is disabled through system programming (→ Remote Programming [810]), then this on site programming cannot be done.

2. **Remote Programming:**

Method	Description
Using a modem (RMT card)	An RMT card must be installed. The floating extension number of the analogue remote maintenance must be assigned (→ Modem Floating Extension Number [811]). PC programming, using a telephone connected in parallel with the modem, can be done in the following ways: • **Direct Access** Dial the DIL/DID/DDI number whose destination is the floating extension number of the analogue remote maintenance. • **Through DISA** Dial the floating extension number of the analogue remote maintenance using the DISA feature. (→ 1.15.6 Direct Inward System Access (DISA)) • **Call Transfer** Call an extension (probably the operator), and request a transfer to the floating extension number of the analogue remote maintenance. (→ 1.11.1 Call Transfer)
Using an external modem (Not RMT card)	An external modem can be used instead of an RMT card to perform the remote programming. An external modem can be connected to the Serial Interface (RS-232C) of the PBX. Connect the modem to an extension port which is assigned as the DIL/DID/DDI/MSN destination or to a trunk directly to connect to the PBX from the PC. An AT command can be sent to the modem automatically when it is connected to the Serial Interface (RS-232C) port. AT commands can be programmed beforehand through system programming to initialise the modem. An AT command can also be sent manually in the PT system programming mode (→ External Modem Control [801]).
Using an ISDN TA interface (64 kbps) through an ISDN Trunk	The floating extension number of the ISDN remote maintenance must be assigned (→ ISDN Remote Floating Extension Number [812]), and dial the DIL/DID/DDI/MSN number whose destination is the floating extension number of the ISDN remote maintenance. The RMT card is not required for this method. This method is available only when an user-supplied ISDN TA that supports CAPI is used.

Conditions

- Access to system programming is allowed only one at a time.
- **Programmer Code and Password for System Programming**
  To access system programming, a programmer code and its password must be entered. There are three programmer codes, which are set at the factory. Each programmer code has a unique password which is also set at the factory.
  Programmer codes can be changed using the Maintenance Console, by selecting "Programmer Code Change" from the "Tool" menu. Passwords can be changed through system programming. If the wrong password is entered a preprogrammed number of times (Default: None [no limitation]) remotely, access will be locked and can only be unlocked by enabling remote access through system programming (→ Remote Programming [810]).

Programmer Code	Password	Description
Installer Level Programmer Code	**System Password for Installer**	All system programming is accessible.
Administrator Level Programmer Code	**System Password for Administrator—for PC Programming**	Permitted system programming is accessible. Each system programming can be determined whether to be accessed by the administrator.
User Level Programmer Code	**System Password for User—for PC Programming**	Permitted system programming is accessible. Each system programming can be determined whether to be accessed by the end users.

- **Remote Programming Limitation**
  It is possible to disable remote access through system programming (→ Remote Programming [810]).
- If an extension user sets the Walking Extension feature while the extension data (e.g., Headset mode) is programmed through PC programming, some extension data for the user may not be transferred properly.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.3.1 MPR Card

2.3.2 RMT Card

2.6.6 CTI-LINK Card

3.1 Overview — KX-TDA100/KX-TDA200

3.2 Connection — KX-TDA100/KX-TDA200

3.3 Installation of the KX-TDA Maintenance Console — KX-TDA100/KX-TDA200

<KX-TDA30>

6.5.6 RMT Card

7.1 Overview — KX-TDA30

7.2 Connection — KX-TDA30

7.3 Installation of the KX-TDA30 Maintenance Console

Feature Guide References

1.1.1.2 Direct In Line (DIL)

1.1.1.3 Direct Inward Dialling (DID)/Direct Dialling In (DDI)

1.1.1.4 Multiple Subscriber Number (MSN) Ringing Service

3.2 PC Programming

2.3.2 PT Programming

Description

A PT user can perform the following programming:

a) **Personal Programming:** Customising the extension according to his needs.

b) **System Programming:** Customising the PBX according to organisational needs.

c) **Manager Programming:** Customising specified frequently changing items (e.g., Charge Management and Remote Extension Lock).

Conditions

- COS programming determines what programming can be performed (→ Programming Mode Limitation [516]):
  - System programming and personal programming
  - Personal programming only
  - No programming
- The extension which is connected to the lowest numbered extension port can perform both personal programming and system programming regardless of the COS.
- The extension(s) assigned as the manager COS can perform manager programming.
- During programming, the PT is considered to be busy.
- Access to system programming and manager programming is allowed only one at a time. However, one PBX supports up to 16 simultaneous programmers (one system programmer + 15 personal programmers, one manager programmer + 15 personal programmers, or 16 personal programmers).
- **System Programming Password Level**
  To access system programming, a valid password must be entered. There are two types of passwords, which are set at the factory, and can be changed through system programming.

Type	Description
System Password for Administrator—for PT Programming	All system programming available for PT programming is accessible (→ System Password for Administrator—for PT Programming [110]).
System Password for User—for PT Programming	Permitted system programming is accessible (→ System Password for User—for PT Programming [111]). Each access to system programming can be controlled.

- **Personal Programming Data Default Set**
  A user can return all of the items programmed on the telephone to default.

Feature Guide References

2.2.6 Manager Features

3.3 PT Programming

User Manual References

User Manual

3.1 Customising Your Phone (Personal Programming)

3.2 Manager Programming

3.3 Customising Your System (System Programming)

2.3.3 Quick Setup

Description

It is possible to set up the basic PBX parameters using a personal computer (PC). When a PC accesses the PBX for the first time with the Installer Level Programmer Code (→ 2.3.1 PC Programming), Quick Setup display will appear automatically. The following items can be programmed as desired:

Item	Parameter	Description
Date & Time Set	Year/Month/ Date/Hour/ Minute	The date and time set on the PC will be used.
System Password for Installer	4–10 characters	Enter the system password for installer.
Operator & Manager	Extension Number	Assign the PBX operator to all time modes (day/lunch/ break/night) (→ 2.2.5 Operator Features). The extension assigned as the PBX operators is automatically allowed to perform manager operation (→ 2.2.6 Manager Features).
Flexible Numbering Type	1. Pattern 1 (with ✳) 2. Pattern 2 (without ✳)	If "Pattern 1 (with ✳)" is selected, "✳" must prefix all feature numbers (except access numbers) when an extension user wants to use a feature. **[Example]** Call Pickup feature number **Pattern 1 (with ✳):** ✳41 **Pattern 2 (without ✳):** 41 For default of the flexible numbers, refer to the [Flexible Numbering Table (available while a dial tone is heard)] (→ 2.3.5 Flexible Numbering/Fixed Numbering).
Operator/ Local Access	1. 0/9 2. 9/0	The feature numbers for Operator Call and Idle Line Access can be selected. They will be "0" or "9".
Remote Maintenance Number	Remote Maintenance Dial Number	Enter the complete telephone number of the PBX (including the country code). When necessary, this number will be used to access the PBX from a remote location for maintenance purposes.

2.3.4 Automatic Setup

Description

There are two automatic setup features as follows:

1) Automatic ISDN Configuration

2) Automatic Time Adjustment

1. Automatic ISDN Configuration

The ISDN (BRI) port configuration can be set automatically through system programming. The following items will be set by making and receiving a call using the subscriber number assigned for each ISDN (BRI) port:

a) L1 Active Mode

b) L2 Data Link Mode

c) Access Mode (Point-to Point/Point-to-Multipoint)

d) TEI Assignment Mode (Fix 00–63/Automatic)

2. Automatic Time Adjustment

It is possible to adjust the PBX clock automatically in the following two ways:

a) Summer Time (Daylight Saving Time) Setting:

The start and end dates of the summer time can be programmed. The PBX clock will be adjusted (one hour forward or backward) at 2:00AM of the programmed date, if enabled through system programming. It means 2:00 AM will become 3:00 AM on the start date of the summer time, and 2:00 AM will become 1:00 AM on the end date.

Note

If the Timed Reminder (Wake-up call) is set;

– On the summer time start date, the setting between 2:00 AM and 3:00 AM will not happen.
– On the summer time end date, the setting between 1:00 AM and 2:00 AM will ring twice.

b) Time Information from Telephone Company:

Time information can be received on the following calls:

- An incoming or outgoing call through an ISDN line
- An incoming call through an analogue line with Caller ID which includes the time information.

The PBX clock will be adjusted everyday with the first call after 3:05 AM, if enabled through system programming.

Note

If the Timed Reminder (Wake-up call) is set, the setting will not happen or will ring twice depending on the adjustment.

Conditions

- SMDR will record the call information using the PBX clock so that the recording time will be overlapped at the end of summer time. (→ 1.24.1 Station Message Detail Recording (SMDR))

Feature Guide References

1.19.1.1 Integrated Services Digital Network (ISDN)—SUMMARY

1.25.4 Timed Reminder

2.3.5 Flexible Numbering/Fixed Numbering

Description

To dial another extension user or to access PBX features, the access numbers (extension numbers or feature numbers) are required.
There are three types of numbering plans:

1) Flexible Numbering (available while a dial tone is heard)

2) Flexible Numbering (available while busy, DND, or ringback tone is heard)

3) Fixed Numbering (available while dialling or talking)

1. Flexible Numbering (available while a dial tone is heard)
Extension numbers and feature numbers which are available while a dial tone is heard can be customised for easy use (→ Flexible Numbering [100]). The numbers must not conflict. It is also possible to use default (Pattern 1 or Pattern 2) shown in the following table:

a) Extension Numbers (Extension Numbering Scheme 1 through 32): Up to four-digit number (up to two-digit leading number consisting of "0 through 9") + additional digits (up to two digits, default: two digits)

[Example]
If the number "3" as a leading number for the 1st extension numbering scheme and the "2" as an additional digit have been programmed, the extension number 300 through 399 are available.

b) Feature Numbers: Up to four-digit number consisting of "0 through 9", "✳", and "#"

c) Other PBX Extension Numbers (Other PBX Extension Number [TIE] -1 through 16): Up to three-digit number consisting of "0 through 9", "✳", and "#"

[Flexible Numbering Table (available while a dial tone is heard)]

Feature	Default			
	Pattern 1 (with ✳)			Pattern 2 (without ✳)
	①*1	②*2	③*3	
Extension Numbering Scheme 1—Leading Number	**2**	**2**	**1**	**1**
Extension Numbering Scheme 2—Leading Number	**3**	**3**	**2**	**2**
Extension Numbering Scheme 3—Leading Number	**4**	**None**	**3**	**None**
Extension Numbering Scheme 4—Leading Number	**5**	**None**	**4**	**None**
Extension Numbering Scheme 5—Leading Number	**6**	**None**	**5**	**None**
Extension Numbering Scheme 6—Leading Number	**None**	**None**	**6**	**None**

Feature	Default	
	Pattern 1 (with ✳)	Pattern 2 (without ✳)
Extension Numbering Scheme 7–32—Leading Number	None	None
Operator Call	9 / 0	9 / 0
Idle Line Access (Local Access)	0 / 9*4	0 / 9
Trunk Group Access	8	8
TIE Line Access	7	None
Redial	#	#
Speed Dialling—System/Personal	✳✳	✳
Personal Speed Dialling—Programming	✳30	30
Doorphone Call	✳31	31
Group Paging	✳33	33
External BGM on/off	✳35	35
Outgoing Message (OGM) playback/record/clear	✳36	36
S-CO Line Access	✳37	37
Parallel Telephone Mode set/cancel	✳39	39
Group Call Pickup	✳40	40
Directed Call Pickup	✳41	41
TAFAS—Calls through an External Pager	✳42	42
Group Paging answer	✳43	43
Automatic Callback Busy cancel/CCBS cancel	✳46	46
User Remote Operation/Walking COS/Verified Code Entry	✳47	47
Wireless XDP Parallel Mode set/cancel	✳48	48
Account Code Entry	✳49	49
Call Hold/Call Hold Retrieve	✳50	50
Call Hold Retrieve—Specified with a Holding Extension Number	✳51	51
Call Park/Call Park Retrieve	✳52	52
Call Hold Retrieve—Specified with a Held Trunk Number	✳53	53
Door Open	✳55	55

Feature	Default	
	Pattern 1 (with ✳)	Pattern 2 (without ✳)
External Feature Access	✳60	60
ISDN Hold	✳62	62
COLR set/cancel	✳7✳0	7✳0
CLIR set/cancel	✳7✳1	7✳1
Switch CLIP/COLP of the Trunk/Extension	✳7✳2	7✳2
MCID	✳7✳3	7✳3
ISDN-FWD set/cancel/confirm	✳7✳5	7✳5
Message Waiting set/cancel/callback	✳70	70
FWD/DND set/cancel—Both	✳710	710
FWD/DND set/cancel—External	✳711	711
FWD/DND set/cancel—Internal	✳712	712
FWD/DND No Answer Timer set	✳713	713
Group FWD set/cancel—Both	✳714	714
Group FWD set/cancel—External	✳715	715
Group FWD set/cancel—Internal	✳716	716
Call Pickup Deny set/cancel	✳720	720
Paging Deny set/cancel	✳721	721
Walking Extension	✳727	727
Data Line Security set/cancel	✳730	730
Call Waiting for Intercom Calls set/cancel	✳731	731
Call Waiting for Trunk Calls (including doorphone calls, calls for an incoming call distribution group) set/cancel	✳732	732
Executive Busy Override Deny set/cancel	✳733	733
Not Ready Mode on/off	✳735	735
Log-in/Log-out	✳736	736
Incoming Call Queue Monitor	✳739	739
Hot Line programme/set/cancel	✳740	740
Absent Message set/cancel	✳750	750
BGM set/cancel	✳751	751

Feature	Default	
	Pattern 1 (with ✳)	Pattern 2 (without ✳)
Timed Reminder set/cancel	✳760	760
Extension Lock set/cancel	✳77	77
Time Service Switch	✳780	780
Remote Extension Lock off	✳782	782
Remote Extension Lock on	✳783	783
Extension Feature Clear	✳790	790
Extension Personal Identification Number (PIN) set/cancel	✳799	799
Dial Information (CTI)	None	None
Other PBX Extension Number (TIE) 1–16	None	None
Quick Dialling 1–80	None	None
KX-T7710 Message Button	✳702	702
KX-T7710 One-touch Dialling 1–8	None	None

*1: For users in Germany, France, Belgium, Norway, Sweden, Denmark, Finland, Portugal, Italy, and New Zealand
*2: For users in the United Kingdom
*3: For users in other countries/areas
*4: For users in New Zealand, the default value of Idle Line Access (Local Access) is "1".

2. Flexible Numbering (available while busy, DND, or ringback tone is heard)
Feature numbers which are available while busy, DND, or ringback tone is heard can be customised for easy use. The numbers should be one digit ("0 through 9" , "✳", or "#") and must not conflict. For default, refer to the following table:

[Flexible Numbering Table (available while busy, DND, or ringback tone is heard)]

Feature	Default
Call Waiting/DND Override	1 or 2*
Executive Busy Override	3
Message Waiting set	4
Call Monitor	5
Automatic Callback Busy/CCBS	6
Alternate Calling—Ring/Voice	✳

*: To use Call Waiting/DND Override, both "1" and "2" are available by default.

3. **Fixed Numbering (available while dialling or talking)**
The features which are available while dialling or talking have fixed numbers as shown in the following table:

[Fixed Numbering Table (available while dialling or talking)]

Feature	Fixed Numbering
Pulse to Tone Conversion	✳
Conference	**3**
Door Open	**5**

Conditions

- The PBX has default for numbers.
- The following are examples of feature number conflicts: 1 and 11, 0 and 00, 2 and 21, 10 and 101, 32 and 321, etc.
- **Feature number + Additional number (Parameter)**
Some flexible feature numbers require additional digits to make the feature active. For example, to set Call Waiting, the feature number for "Call Waiting" must be followed by "1" and to cancel it, the same feature number should be followed by "0".
- If a feature number includes "✳" or "#", rotary SLT users cannot use it.
- ISDN extension users cannot use the following features:
  - OGM playback/record
  - Call Hold/Call Hold Retrieve (held at its own extension)
  - ISDN Hold
  - MCID
  - Walking Extension
  - Call Waiting
  - Hot Line
  - Timed Reminder
  - Executive Busy Override
  - Call Monitor
  - Automatic Callback Busy/CCBS
- PS users cannot use the following features:
  - Personal Speed Dialling
  - OGM playback/record
  - S-CO Line Access
  - Parallel Telephone Mode set/cancel
  - Walking Extension
  - BGM set/cancel
  - Timed Reminder

User Manual References

User Manual

4.2.1 Feature Number Table

2.3.6 Floating Extension

Description

Virtual extension numbers can be assigned to resources to make them appear as extensions. These numbers are defined as floating extension numbers and can be assigned as a destination of incoming calls etc.

This feature is also known as Floating Station.

Resource		Description
Device	**External Pager**	Used as the destination for TAFAS feature (→ External Pager Floating Extension Number [700]). (→ 1.15.3 Trunk Answer From Any Station (TAFAS))
	Outgoing Message (OGM)	Used to send messages for DISA feature (→ Outgoing Message (OGM) Floating Extension Number [730]). (→ 1.15.6 Direct Inward System Access (DISA))
	Analogue Remote Maintenance	Used to access the PC programming mode through a modem on a personal computer (→ Modem Floating Extension Number [811]).
	ISDN Remote Maintenance	Used to access the PC programming mode through the ISDN interface on a personal computer (→ ISDN Remote Floating Extension Number [812]).
Group	**Incoming Call Distribution Group**	Used to call an incoming call distribution group (→ Incoming Call Distribution Group Floating Extension Number [622]). (→ 1.2.2.1 Incoming Call Distribution Group Features—SUMMARY)
	PS Ring Group	Used to call a PS ring group. (→ 1.23.2 PS Ring Group)
	VM (DPT) Group	Used to call a VM (DPT) group (→ VM Group Floating Extension Number [660]).
	VM (DTMF) Group	Used to call a VM (DTMF) group.

Conditions

- It is possible to give names to floating extension numbers (→ Incoming Call Distribution Group Name [623], Outgoing Message (OGM) Name [731]).

Feature Guide References

1.22.1 Voice Mail (VM) Group

2.3.1 PC Programming

2.3.7 Software Upgrading

Description

It is possible to upload software from a personal computer to the PBX to upgrade the following:

Data File	Storage Area
Main Processing (MPR) software data	Operating system data area on either the MPR card of the KX-TDA100/KX-TDA200, or the Main Board of the KX-TDA30.
Default data of system programming for each country/ area	Country/area data area on the MPR card or Main Board
Language data for PT display	Each language data area on the MPR card or Main Board. There are two types: **Type 1:** PT display except for PT system programming (→ 2.3.2 PT Programming) (five languages max.) **Type 2:** PT system programming data (one language max.)
LPR (software on a slot card) software data (KX-TDA100/ KX-TDA200 only)	Flash ROM on the slot card (e.g., BRI8)
Cell Station (CS) software data	Flash ROM on the CS

Conditions

- The software version of the MPR can be confirmed through system programming (→ Main Processing (MPR) Software Version Reference [190]).

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.3.1 MPR Card

2.4 Fault Recovery/Diagnostics

2.4.1 Power Failure Transfer

Description

When the power supply to the PBX fails, specific SLTs and/or ISDN terminal devices are automatically connected to specific trunks (**Power Failure Connections**). The PBX will switch from normal operation to the Power Failure Connections, and all existing conversations will be disconnected.
Only the trunks handled by Power Failure Connections can be used during a power failure.

Conditions

[General]

- Only trunk calls can be made during a power failure. All other features do not work.
- **DC Power Source**
  If the DC power is available from backup batteries, even if the AC power fails, the PBX will remain fully operational.

[KX-TDA100/KX-TDA200]

- The analogue trunk cards and extension cards and the number of ports which can be used for Power Failure Connections are as follows:
  Trunk Cards: LCOT16 (four ports) and LCOT8 (two ports)
  Extension Cards: MSLC16 (four ports), SLC16 (four ports), DHLC8 (two ports) and SLC8 (two ports).
  To each trunk card, connect only one extension card.

  Note

  Power Failure Connections between analogue trunk cards and extension cards should be done as per system programming so that conversation is maintained when the power is restored and the MPR is recovering.

- The BRI8 (one port) and BRI4 (one port) cards can also be used for Power Failure Connections. For details about Power Failure Connections, refer to the Installation Manual.

[KX-TDA30]

- Ports 1 and 2 of the LCOT4 card installed in the least slot number and ports 1 and 2 of the super hybrid ports can be used for Power Failure Connections.

  Note

  The Power Failure Connections between the LCOT4 card and super hybrid ports are automatically kept so that the conversation is maintained when the power is restored and the Main Board is recovering.

- The BRI2 (one port) card can also be used for Power Failure Connections. For details about Power Failure Connections, refer to the Installation Manual.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

2.2.6 Backup Batteries Connection

2.12.1 Power Failure Connections

<KX-TDA30>

6.2.6 Backup Batteries Connection

6.11.1 Power Failure Connections

2.4.2 Power Failure Restart

Description

When turning the electricity back on, the PBX restarts the stored data automatically and the PBX will record the event (System Restart) in the error log.

Conditions

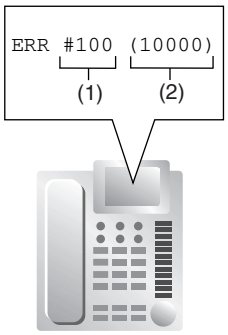
- In the event of a power failure, PBX memory is protected by a factory-provided lithium battery. There is no memory loss except the memories of Automatic Callback Busy (Camp-on) (→ 1.7.1 Automatic Callback Busy (Camp-on)) and Call Park (→ 1.12.2 Call Park).

2.4.3 Local Alarm Information

Description

When a PBX error occurs and the PBX detects it, the System Alarm button light on the PT of an extension, which is allowed to use this feature through system programming (a maximum of two extensions per PBX), turns on red. Pressing the button will show the error number on the display. If multiple errors occur, the error number will be displayed in order of highest priority to lowest. The System Alarm button light turns off automatically after all error numbers have been displayed.
For details about the errors and their solutions, refer to the Installation Manual.

[Error Example]



[Explanation]

Number in the Example	Item	Description
(1)	**Error Code**	Shows three-digit error code.
(2)	**Sub Code**	Shows five-digit sub code (XYYZZ). **X:** Cabinet number (1–4) **YY:** Slot number (00–11) **ZZ:** Physical port number (01–16)

Conditions

- **System Alarm Button**
  A flexible button can be customised as the System Alarm button.
- If an extension allowed to utilise this feature uses a PC Console or PC Phone, the alarm information will be displayed on the PC.
- If an extension allowed to utilise this feature uses a PC Console or PC Phone, the alarm information will be sent to the preprogrammed party via the PC Console or PC Phone. The PC Console and PC Phone use e-mail to send the information.
- The alarm information will be recorded on the SMDR, if enabled through system programming.
- The PBX can be automatically diagnosed at a preprogrammed time everyday.

Installation Manual References

Installation Manual

<KX-TDA100/KX-TDA200>

4.1.5 Troubleshooting by Error Log

<KX-TDA30>

8.1.5 Troubleshooting by Error Log

Feature Guide References

1.18.2 Flexible Buttons

1.24.1 Station Message Detail Recording (SMDR)

Section 3

Programming Instructions

3.1 Introduction

3.1.1 Introduction

These Programming Instructions are designed to serve as an overall system programming reference for the Panasonic Hybrid IP-PBX. The PBX has a default setting. You can change the default to meet your requirements. System programming controls the PBX features described in this Feature Guide. It allows the PBX to be customised to the requirements. Only one person can programme at a time. Another person trying to enter programming mode will be rejected.

Ways to Programme

There are two programming methods:

- **PC (Personal Computer) Programming**
  All system programming can be performed. PC programming is described in Section 3.2 PC Programming.
- **PT (Proprietary Telephone) Programming**
  A subset of system programming can be performed. PT programming is described in Section 3.3 PT Programming. The extension user can programme by entering 3 digits programming numbers from the PT.

Password Security

A password is required to perform programming for security purposes. Do not disclose the password. This will avoid unauthorised access and possible dial through fraud.

Warning to the Administrator regarding the system password

1. Please inform the customer of the importance of the password and the possible dangers.
2. Please maintain the secrecy of the password. This will avoid unauthorised access and possible dial through fraud.
3. Please change the password periodically.
4. We strongly recommend that a password of 10 digits is used for maximum protection against hackers.
5. If the system password is forgotten, you can examine the backup of the system programming. Therefore, please keep your backup secure to avoid unauthorised access.

- If you have a backup of the system data, you can find the password by loading the backup system data onto the PC and check the password using the programming tool. For how to back up the system data, refer to Section 3.3.4 Hybrid IP-PBX Maintenance (for KX-TDA100/KX-TDA200) or 7.3.4 Hybrid IP-PBX Maintenance (for KX-TDA30) in the Installation Manual.
- If you do not have a backup system data, you have to set the PBX to the factory default and reprogramme it. Therefore, we recommend that you back up the system data.

3.2 PC Programming

3.2.1 Installing and Starting the Maintenance Console

To programme and administer the PBX by personal computer (PC), you need to install the Maintenance Console (KX-TDA Maintenance Console for the KX-TDA100/KX-TDA200 or KX-TDA30 Maintenance Console for the KX-TDA30) onto the computer.
This section briefly describes how to install and start the Maintenance Console when the PC and the PBX are connected by USB cable. The screenshots shown in the installation procedure are based on the KX-TDA Maintenance Console.

System Requirements

Operating System

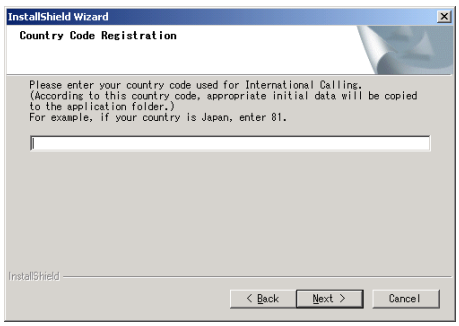
- Microsoft® Windows® 98 SE, Windows Me, Windows 2000, or Windows XP

Hardware

- CPU: Intel® Pentium® 133 MHz or better microprocessor
- RAM: at least 64 megabytes (MB) of free RAM (128 MB recommended)
- HDD: at least 100 MB of hard disc space

Installing the Maintenance Console and Selecting the Appropriate Country/Area Data

1. **a.** Save the setup file of the Maintenance Console on your PC.
   b. Double-click the icon to execute the setup file.
   c. Follow the instructions of the wizard.


InstallShield Wizard
Country Code Registration
Please enter your country code used for International Calling.
(According to this country code, appropriate initial data will be copied to the application folder.)
For example, if your country is Japan, enter 81.
InstallShield
< Back
Next >
Cancel

2. **a.** Type the appropriate **Country Code**. The Maintenance Console will be installed with the appropriate default data for your country/area.
   b. Click [Next].
   c. Follow the instructions of the wizard.
   d. Click [Finish].
   e. Click [OK].

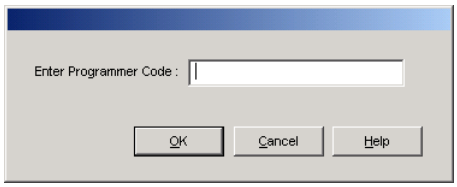
Notes

- To install or uninstall the software into Windows 2000 Professional or Windows XP Professional, the user must be grouped either of "Administrators" or "Power Users".
- To connect the PC to the PBX via USB, the KX-TDA USB driver must have been installed. Follow the instructions of the wizard to install the KX-TDA USB driver.
- The contents and design of the software are subject to change without notice.

Starting the Maintenance Console and Assigning the Basic Items (Quick Setup)

When you start the Maintenance Console with the Installer Level Programmer Code and connect to the PBX for the first time after initialisation (with the factory default setting), Quick Setup will launch automatically. For details about Quick Setup items, refer to Section 2.3.3 Quick Setup.

1. Connect the PC to the PBX with a USB cable.
2. Start the **KX-TDA Maintenance Console** or **KX-TDA30 Maintenance Console** from the start menu.

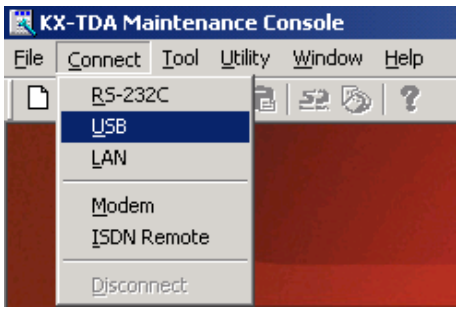


3. Type the Installer Level Programmer Code (default: **1234**), then click [OK].

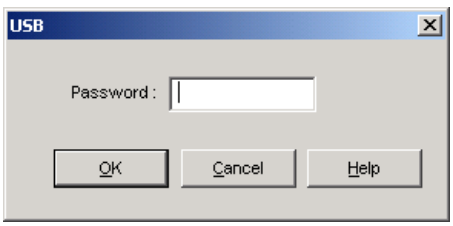
The Programmer Code authorises different programming levels, and the Quick Setup is only available when you start the Maintenance Console with the Installer Level Programmer Code.

Note

There are 2 other Programmer Codes with limited authorisation: Administrator Level (default: **1111**), and User Level (default: none).

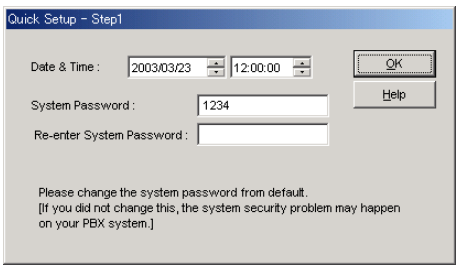


4. Click "**Connect**" → "**USB**" from the menu bar.

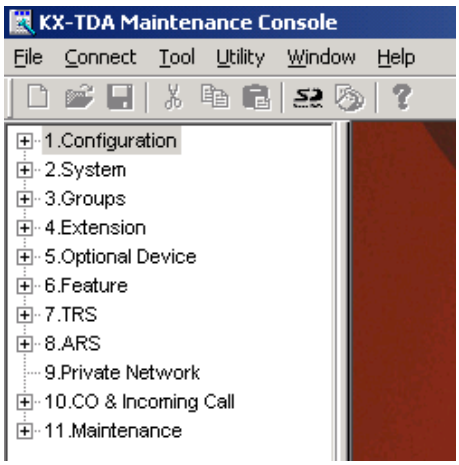


5. Type the system password for installer (default: **1234**), then click [OK] to log-in.

6. **When country/area data do not match:**
   a. Click [OK] to replace the country/area data of the PBX. Replacement may take several minutes to complete.
   b. Follow the procedure described in Section 2.13.1 Starting the Hybrid IP-PBX (for KX-TDA100/KX-TDA200) or 6.12.1 Starting the Hybrid IP-PBX (for KX-TDA30) in the Installation Manual and restart the PBX.
   c. Repeat steps **2** to **4** to restart the Maintenance Console.



7. Follow the instructions of the wizard and assign the basic items (Quick Setup).



The programme menu appears.

Notice

1. During a long programming session, it is highly recommended that you periodically save the system data to the SD Memory Card. You can think of system data as stored in RAM, whereas SD Memory Card as stored on a hard disk. If the PBX undergoes a sudden power failure or system reset for some reason, all the system data in RAM will be lost.
   To save the system data to the SD Memory Card, (1) click the "**SD Memory Backup**" icon before resetting the PBX or turning off the power, or (2) exit the Maintenance Console so that the PBX starts automatically saving the system data.
2. When the PBX is initialised, not all data is taken from the SD Memory Card. The data for present status of extension FWD/DND buttons is taken from battery backup memory in the PBX.

CAUTION

Do not remove the SD Memory Card during access to it. Doing so will damage the system data, and in the worst case, damage the SD Memory Card.

3.3 PT Programming

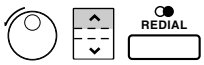
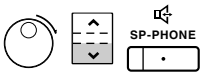
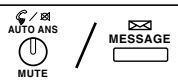
3.3.1 Programming Instructions

Required Telephone/Extension

A PT with multiline display (e.g., KX-T7636 6-line display) is required for the system programming. The extension must be in a COS permitted System Programming or the extension connected to the smallest port number.

System programming allows only one extension user to enter at a time.

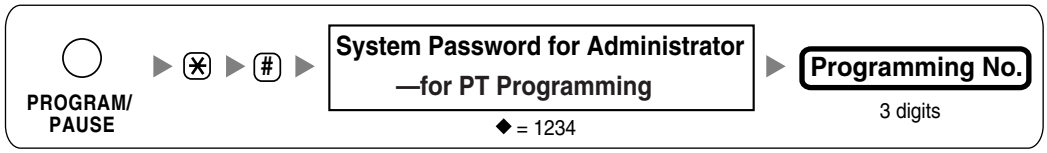
Buttons and Functions

Fixed Buttons		Function
T7600	T7200/T7400/T7500	
		PREVIOUS
		NEXT
		(Page up)
		(Page down)
		ENTER
	—	Back to Previous Menu (CANCEL)
		SHIFT
		END
		CLEAR
		SELECT
		PROGRAM
		FLASH
		SECRET

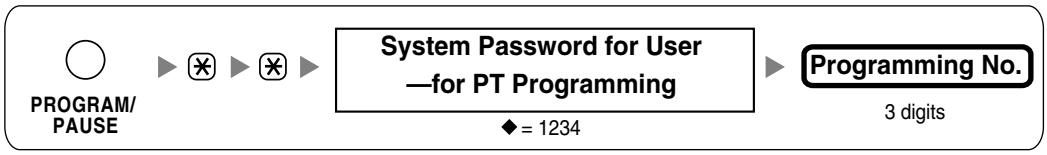
Entering System Programming Mode

System programming enables the extension user to set the time, call destination and all sorts of system conditions within the limits of available programming. The administrator or manager can also set up desired parameters for user programming. It is necessary to enter the relevant password to start programming. Programming has two levels: Administrator Level and User Level.

Administrator Level:
All system programming available for PT programming is accessible.



User Level:
Permitted system programming is accessible. Each access to system programming can be controlled.



Note

◆ means default value throughout this manual.

Programming Structure

Programming Number	Programming Group Title	Description
[0XX]	Basic Programming	Frequently used programming steps.
[1XX]	System Programming	Global system parameters.
[2XX]	Time Programming	Flexible system timers.
[3XX]	TRS/Barring/ARS Programming	TRS/Barring and Automatic Route Selection (ARS).
[4XX]	Trunk Programming	Setting of trunk, trunk group, or BRI line features.
[5XX]	COS Programming	Class of Service parameters.
[6XX]	Extension Programming	Assignment of the extension features
[7XX]	Resource/Interface Programming	Assignment of the various interfaces on the PBX.
[8XX]	SMDR & Maintenance Programming	Setting of SMDR and maintenance features.
[9XX]	Card Programming	The programming used to refer to or delete the card.

Entering Characters

To store a name or message, you can enter the following characters. Those tables show you the characters available on each button.

Table 1 (Standard mode)

Times / Buttons	1	2	3	4	5	6	7	8	9
1	!	?	"	1					
2	A	B	C	a	b	c	2		
3	D	E	F	d	e	f	3		
4	G	H	I	g	h	i	4		
5	J	K	L	j	k	l	5		
6	M	N	O	m	n	o	6		
7	P	Q	R	S	p	q	r	s	7
8	T	U	V	t	u	v	8		
9	W	X	Y	Z	w	x	y	z	9
0	(space)	.	,	'	:	;	0		
*	/	+	–	=	<	>	*		
#	$	%	&	@	(	)	€	£	#

Table 2 (Option mode)

Times / Buttons	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
1	!	?	"	ä	ö	ü	1								
2	A	B	C	a	b	c	À	Á	Â	Ã	Ä	Å	Æ	Ç	2
3	D	E	F	d	e	f	Ð	È	É	Ê	Ë	3			
4	G	H	I	g	h	i	Ì	Í	Î	Ï	4				
5	J	K	L	j	k	l	5								
6	M	N	O	m	n	o	Ñ	Ò	Ó	Ô	Õ	Ö	Ø	Œ	6
7	P	Q	R	S	p	q	r	s	ß	7					
8	T	U	V	t	u	v	Ù	Ú	Û	Ü	8				
9	W	X	Y	Z	w	x	y	z	Ý	9					
0	(space)	.	,	'	:	;	0								
*	/	+	–	=	<	>	*								
#	$	%	&	@	(	)	€	£	#						

Table 2 (Option mode for GR model)

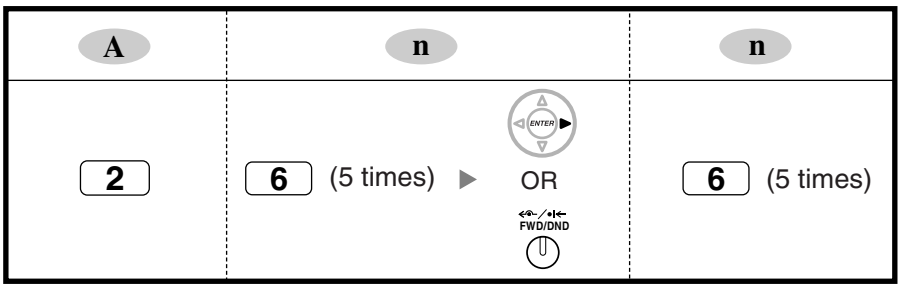
Times / Buttons	1	2	3	4	5	6	7	8	9
1	!	?	"	1					
2	A	B	Γ	2					
3	Δ	E	Z	3					
4	H	Θ	I	4					
5	K	Λ	M	5					
6	N	Ξ	O	6					
7	Π	P	Σ	7					
8	T	Y	Φ	8					
9	X	Ψ	Ω	9					
0	(space)	.	,	'	:	;	0		
*	/	+	–	=	<	>	*		
#	$	%	&	@	(	)	€	£	#

Table 2 (Option mode for RU model)

Times / Buttons	1	2	3	4	5	6	7	8	9	10
1	А	Б	В	!	?	"	1			
2	Г	Д	Е	2						
3	Ж	З	И	Й	3					
4	К	Л	М	4						
5	Н	О	П	5						
6	Р	С	Т	6						
7	У	Ф	Х	7						
8	Ц	Ч	Ш	8						
9	Щ	Ъ	Ы	Ь	9					
0	Э	Ю	Я	(space)	.	,	'	:	;	0
*	/	+	–	=	<	>	*			
#	$	%	&	@	(	)	€	£	#	

[Example of Entering Characters]

To enter "Ann":



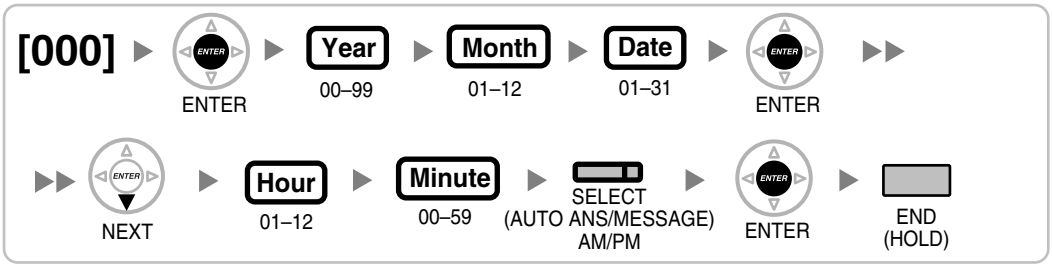
Notes

- To erase the character on the cursor, press CLEAR.
- To move the cursor to the left, press ◀.
- To move the cursor to the right, press ▶.
- If SELECT is pressed, the characters for each button will bring out in reverse order.
- To toggle between Table 1 and Table 2 when using KX-T7600 series, press the leftmost soft button.

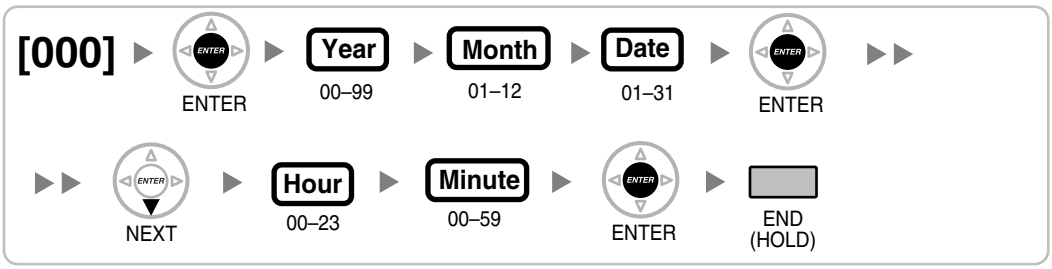
3.3.2 Basic Programming

Date & Time [000]

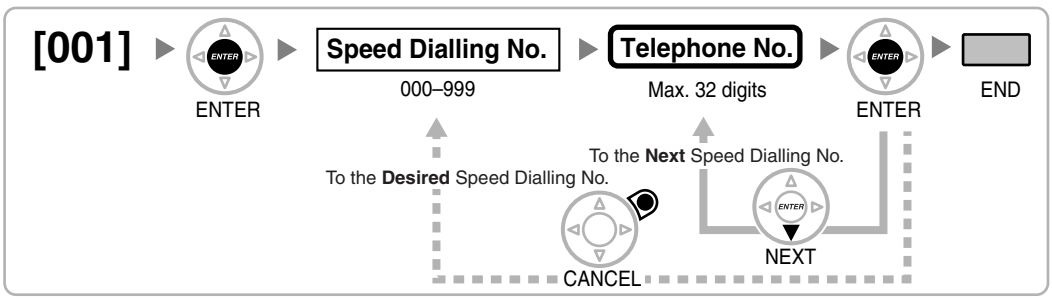
[In 12-hour Format]



[In 24-hour Format]

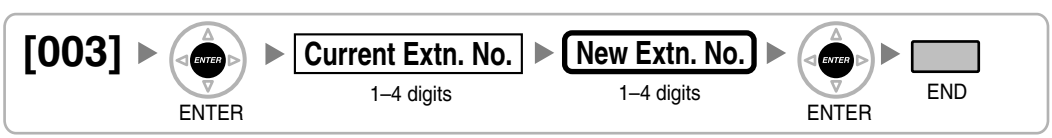


System Speed Dialling Number [001]



System Speed Dialling Name [002]

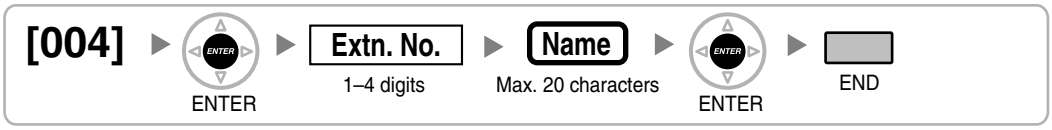
Extension Number [003]



Note

If a PC Phone, PC Console or CTI application software is running on the PC that is connected to the PBX, quit it before this programme.

Extension Name [004]



Extension Personal Identification Number (PIN) [005]

WARNING

When a third party discovers the personal identification number (PIN) (verified code PIN/ extension PIN) of the PBX, there will be a risk that they will make fraudulent phone calls using the telephone line, and the cost will be charged to your own account. In order to avoid this problem, we strongly recommend the following points:

a) Carefully maintain the secrecy of the PIN.

b) Specify a complicated PIN as long and random as you can make it.

c) Change the PIN frequently.

Operator Assignment [006]

Console Paired Telephone [007]

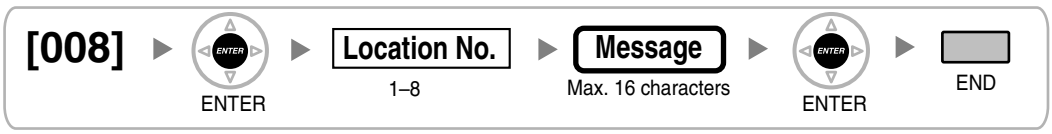
[KX-TDA100/KX-TDA200]

[KX-TDA30]

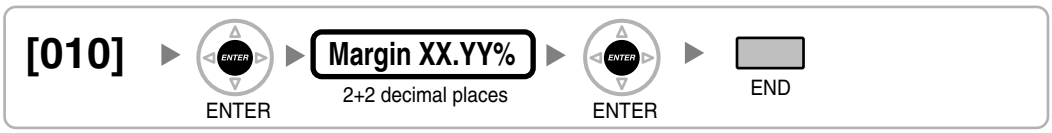
Note

This programme is available after the port connected to the Console is assigned as "Console" in Terminal Device Assignment [601].

Absent Message [008]

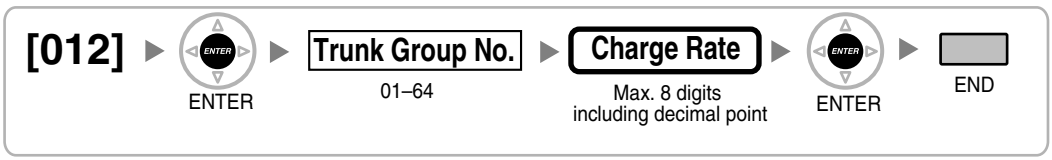


Charge Margin [010]



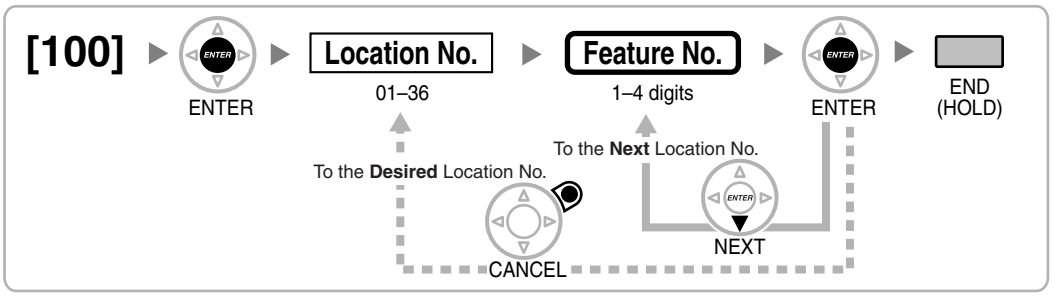
Charge Tax [011]

Charge Rate per Unit [012]

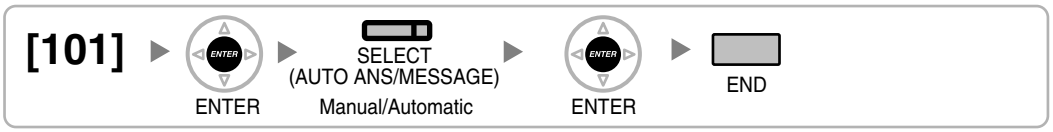


3.3.3 System Programming

Flexible Numbering [100]

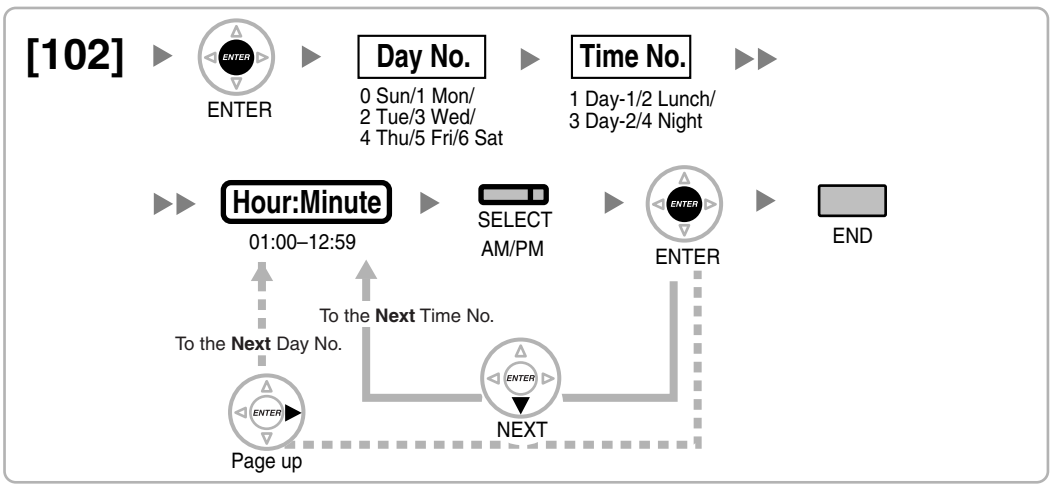


Time Service Switching Mode [101]

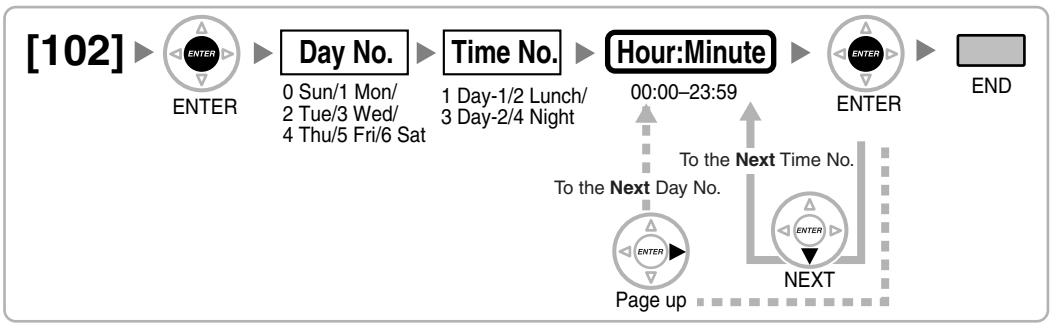


Time Service Starting Time [102]

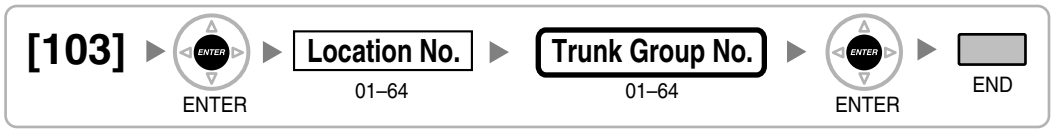
[In 12-hour Format]



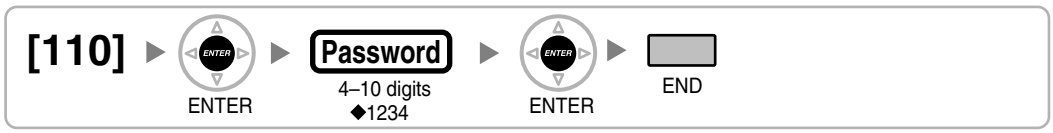
[In 24-hour Format]



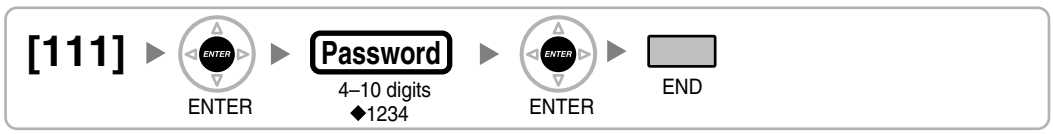
Idle Line Access (Local Access) [103]



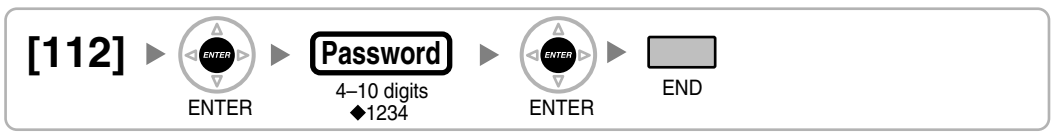
System Password for Administrator—for PT Programming [110]



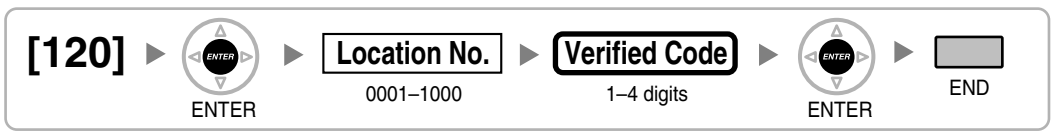
System Password for User—for PT Programming [111]



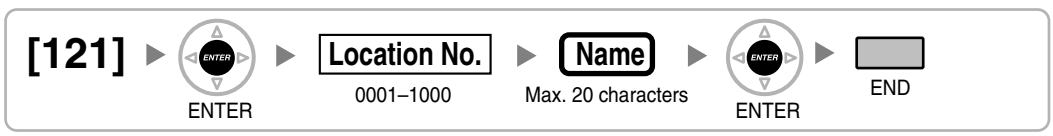
Manager Password [112]



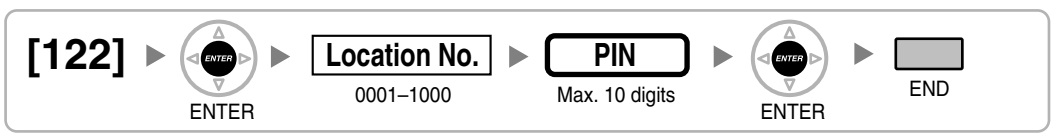
Verified Code [120]



Verified Code Name [121]



Verified Code Personal Identification Number (PIN) [122]



WARNING

When a third party discovers the personal identification number (PIN) (verified code PIN/ extension PIN) of the PBX, there will be a risk that they will make fraudulent phone calls

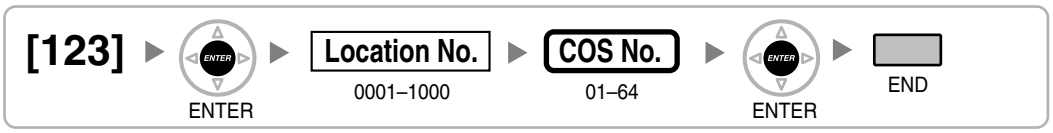
using the telephone line, and the cost will be charged to your own account. In order to avoid this problem, we strongly recommend the following points:

a) Carefully maintain the secrecy of the PIN.

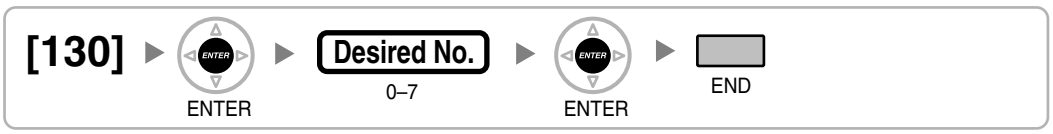
b) Specify a complicated PIN as long and random as you can make it.

c) Change the PIN frequently.

Verified Code COS Number [123]



Decimal Point Position for Currency [130]



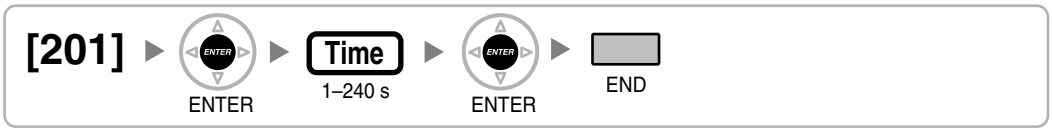
Currency [131]

Main Processing (MPR) Software Version Reference [190]

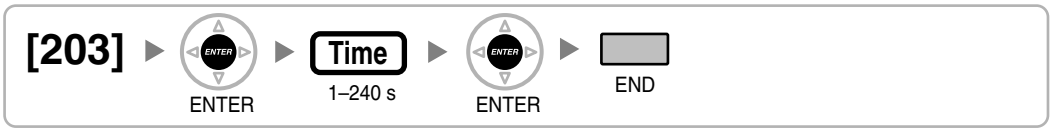
3.3.4 Time Programming

Hold Recall Time [200]

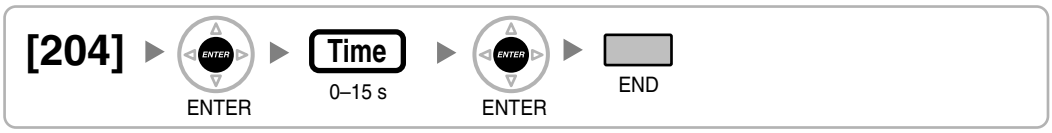
Transfer Recall Time [201]



Intercept Time [203]

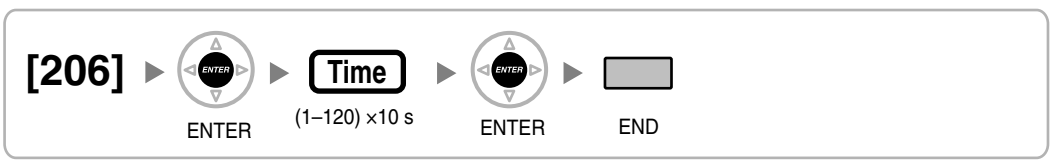


Hot Line Waiting Time [204]

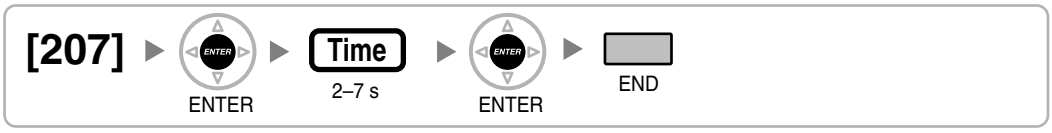


Automatic Redial Repeat Times [205]

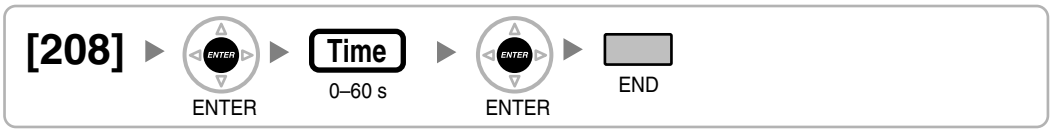
Automatic Redial Interval [206]



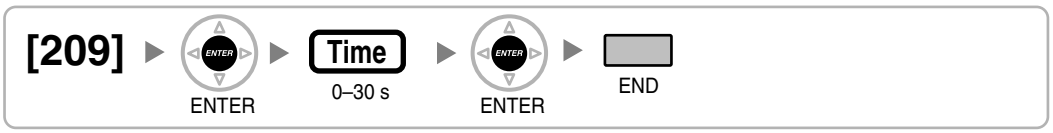
Door Open Duration Time [207]



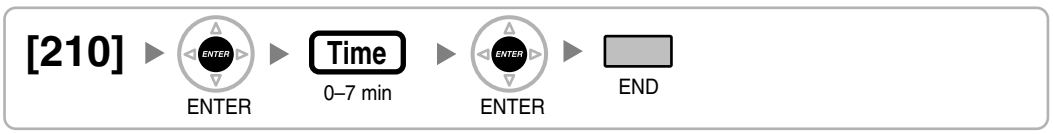
Call Duration Count Starting Time for LCOT [208]



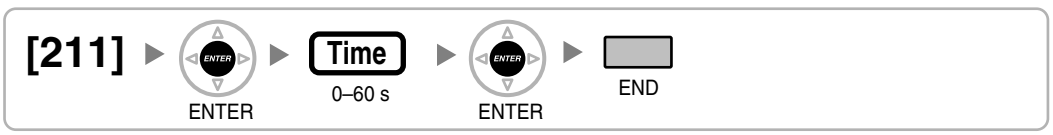
DISA Delayed Answer Time [209]



DISA Trunk-to-Trunk Call Prolong Time [210]

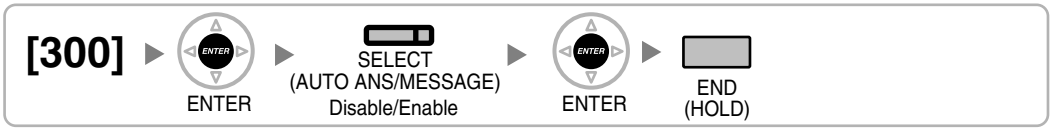


DISA Intercept Time [211]

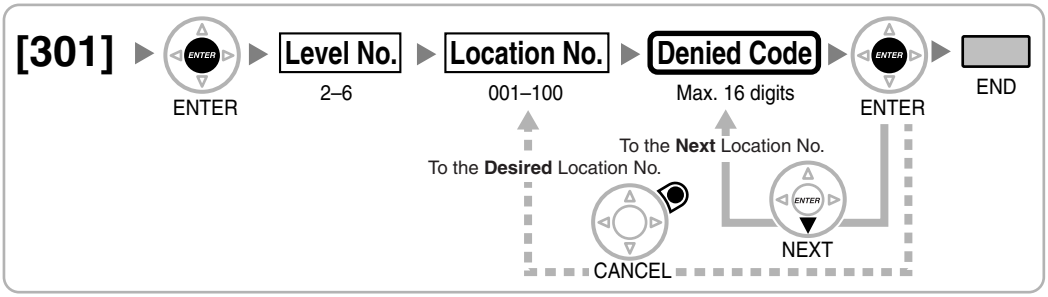


3.3.5 TRS/Barring/ARS Programming

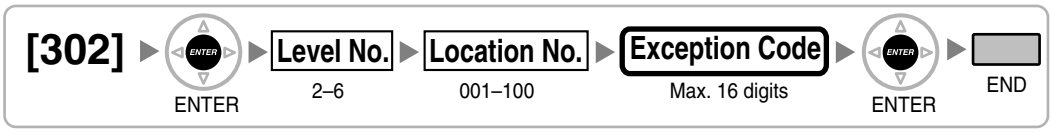
TRS/Barring Override by System Speed Dialling [300]



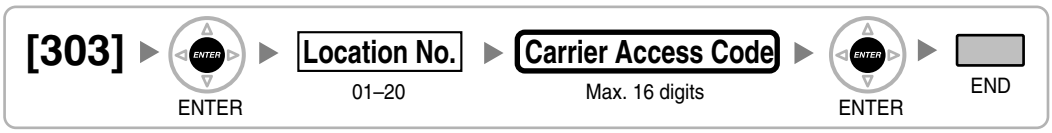
TRS/Barring Denied Code [301]



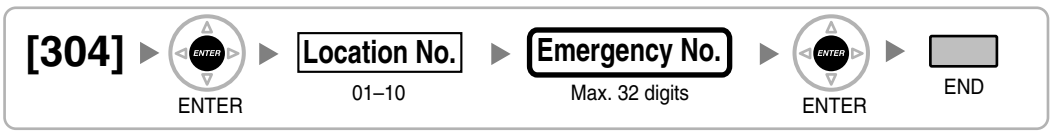
TRS/Barring Exception Code [302]



Special Carrier Access Code [303]

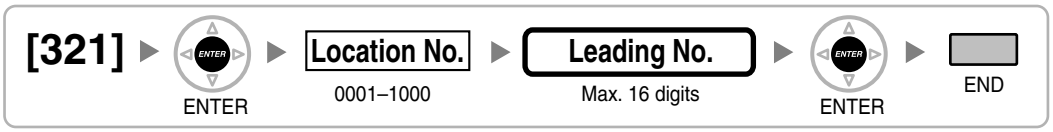


Emergency Number [304]

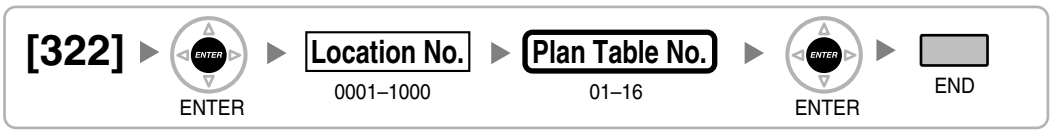


ARS Mode [320]

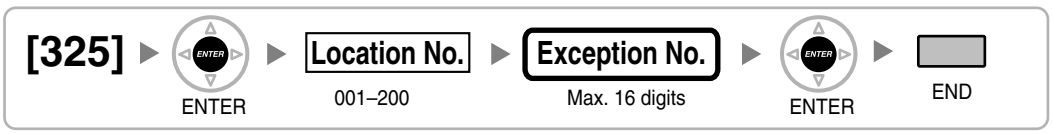
ARS Leading Number [321]



ARS Routing Plan Table Number [322]

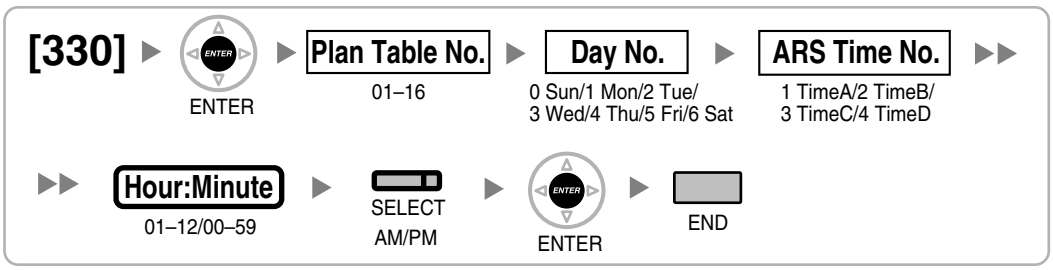


ARS Exception Number [325]

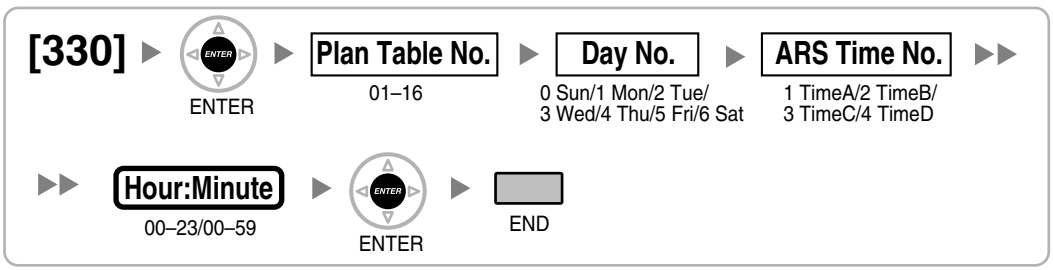


ARS Routing Plan Time Table [330]

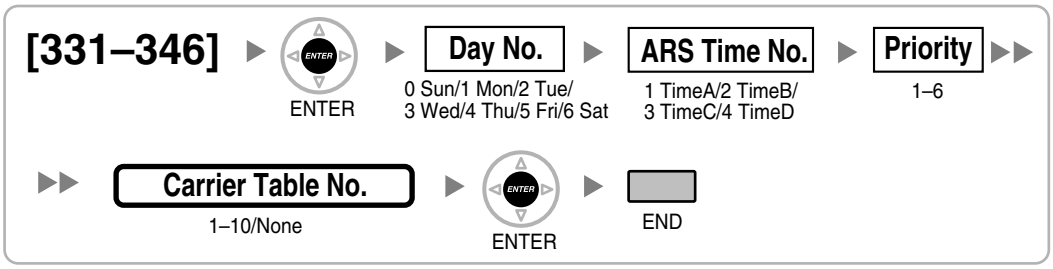
[In 12-hour Format]



[In 24-hour Format]

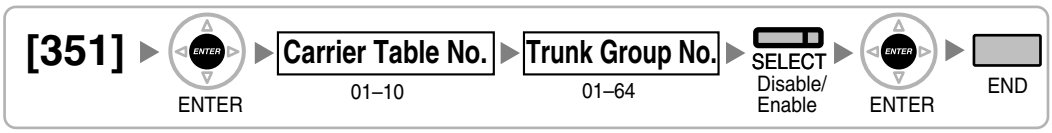


ARS Routing Plan Table (1–16) [331–346]

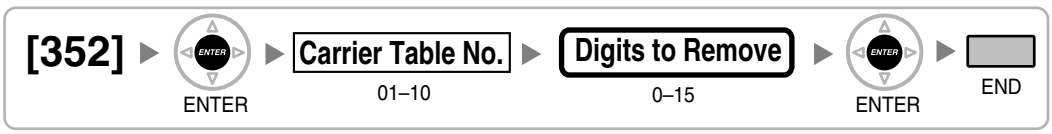


ARS Carrier Name [350]

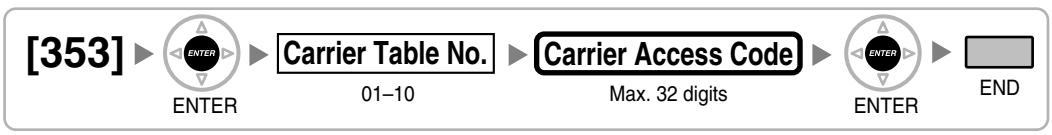
ARS Trunk Group for Carrier Access [351]



ARS Removed Number of Digits for Carrier Access [352]



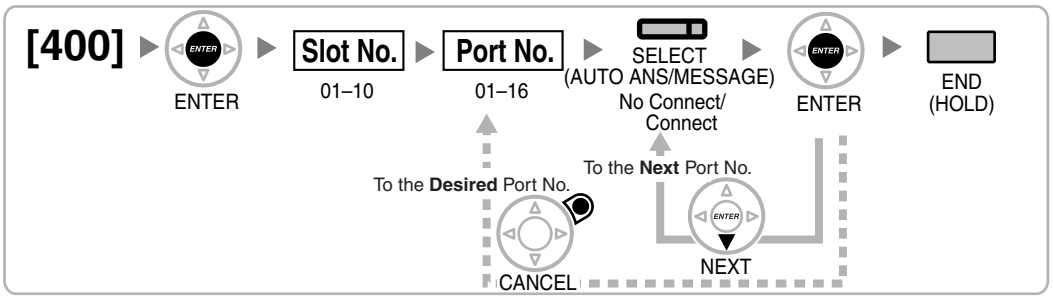
ARS Carrier Access Code [353]



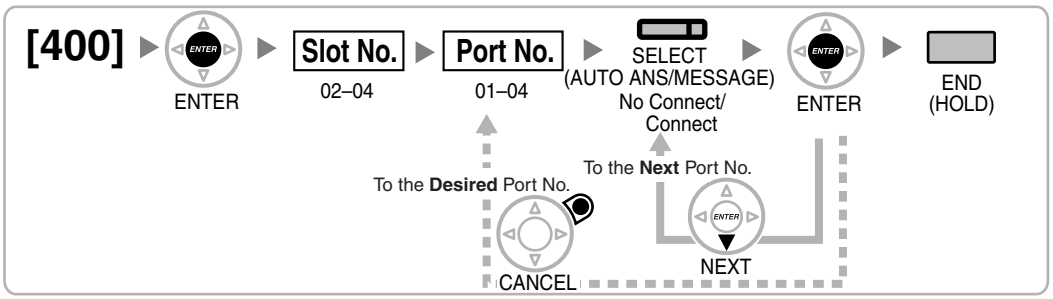
3.3.6 Trunk Programming

LCOT/BRI Trunk Connection [400]

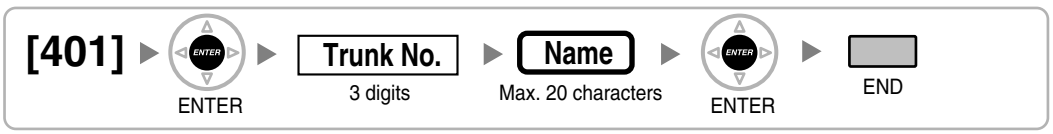
[KX-TDA100/KX-TDA200]



[KX-TDA30]



LCOT/BRI Trunk Name [401]



LCOT/BRI Trunk Group Number [402]

LCOT/BRI Trunk Number Reference [409]

[KX-TDA100/KX-TDA200]

[KX-TDA30]

LCOT Dialling Mode [410]

[KX-TDA100/KX-TDA200]

[KX-TDA30]

LCOT Pulse Rate [411]

[KX-TDA100/KX-TDA200]

[KX-TDA30]

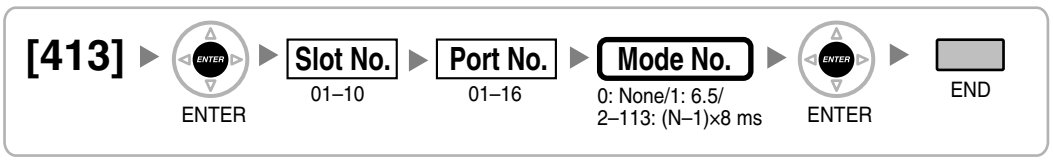
LCOT DTMF Minimum Duration [412]

[KX-TDA100/KX-TDA200]

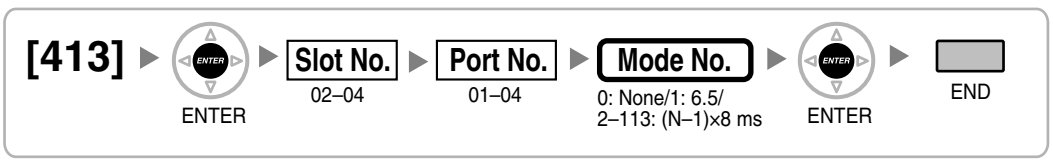
[KX-TDA30]

LCOT CPC Signal Detection Time—Outgoing [413]

[KX-TDA100/KX-TDA200]

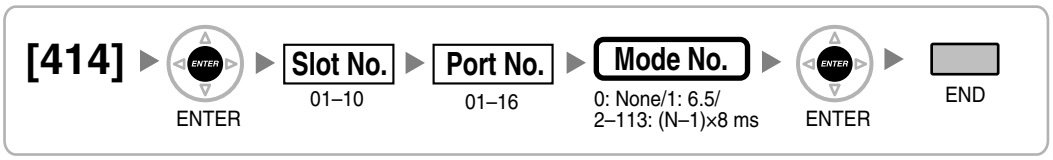


[KX-TDA30]

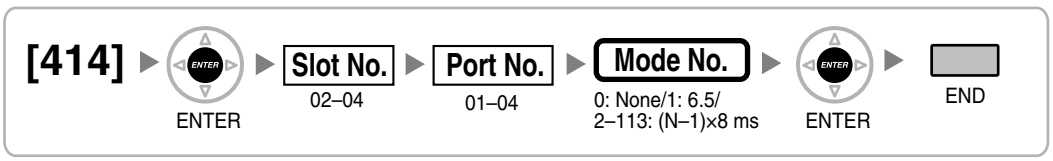


LCOT CPC Signal Detection Time—Incoming [414]

[KX-TDA100/KX-TDA200]

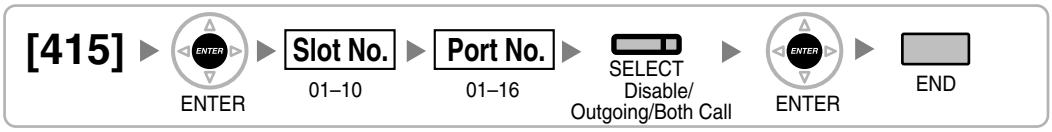


[KX-TDA30]

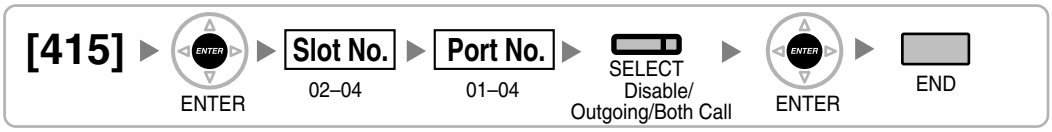


LCOT Reverse Circuit [415]

[KX-TDA100/KX-TDA200]

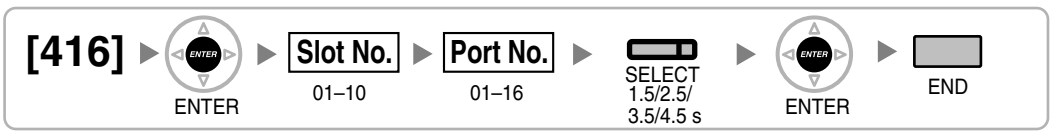


[KX-TDA30]

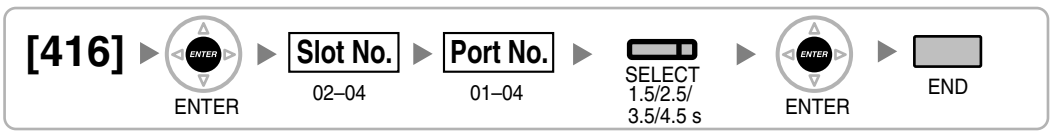


LCOT Pause Time [416]

[KX-TDA100/KX-TDA200]

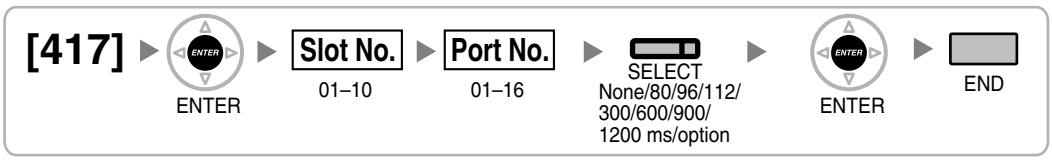


[KX-TDA30]

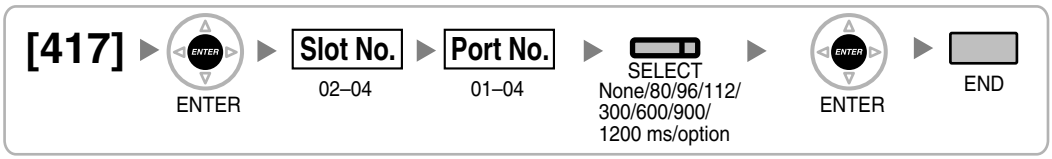


LCOT Flash/Recall Time [417]

[KX-TDA100/KX-TDA200]



[KX-TDA30]



LCOT Disconnect Time [418]

[KX-TDA100/KX-TDA200]

[KX-TDA30]

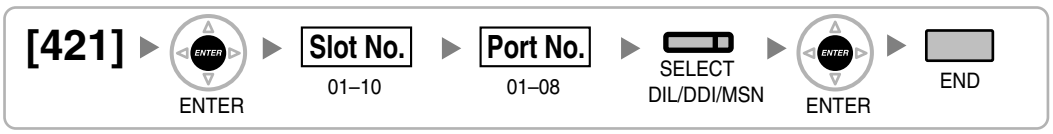
BRI Network Type [420]

[KX-TDA100/KX-TDA200]

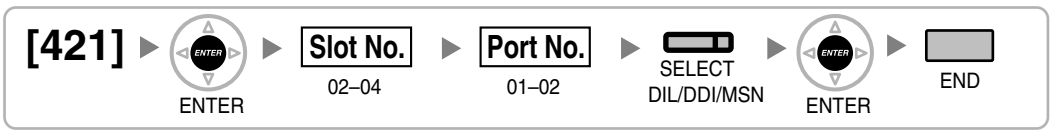
[KX-TDA30]

BRI DIL/DDI/MSN Selection [421]

[KX-TDA100/KX-TDA200]



[KX-TDA30]



BRI Subscriber Number [422]

[KX-TDA100/KX-TDA200]

[KX-TDA30]

BRI Layer 1 Active Mode [424]

[KX-TDA100/KX-TDA200]

[KX-TDA30]

Note

This programme needs Slot Card Reset [902] to enable after setting.

BRI Layer 2 Active Mode [425]

[KX-TDA100/KX-TDA200]

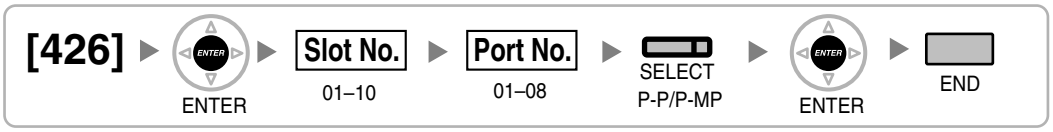
[KX-TDA30]

Note

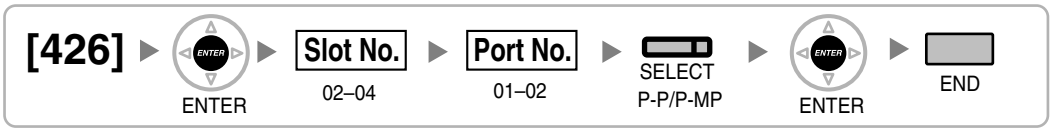
This programme needs Slot Card Reset [902] to enable after setting.

BRI Configuration [426]

[KX-TDA100/KX-TDA200]



[KX-TDA30]

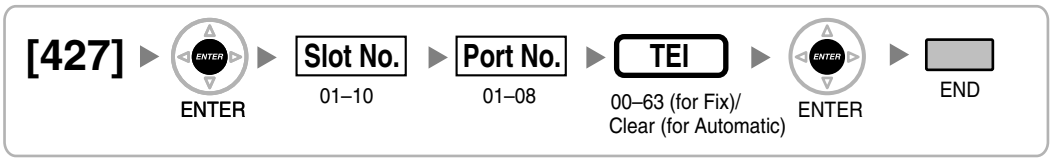


Note

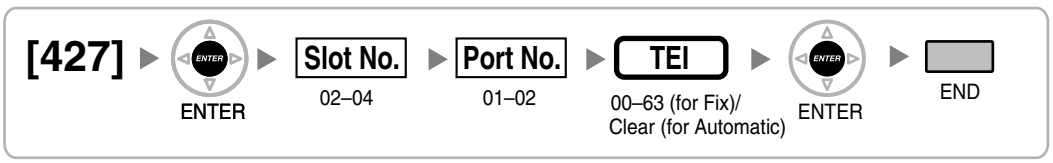
This programme needs Slot Card Reset [902] to enable after setting.

BRI TEI Mode [427]

[KX-TDA100/KX-TDA200]



[KX-TDA30]

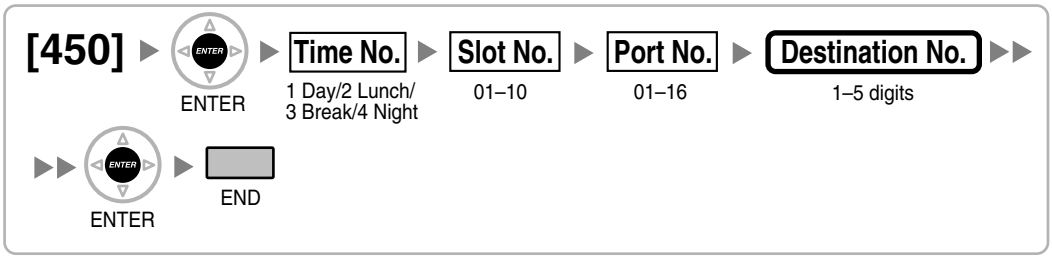


Note

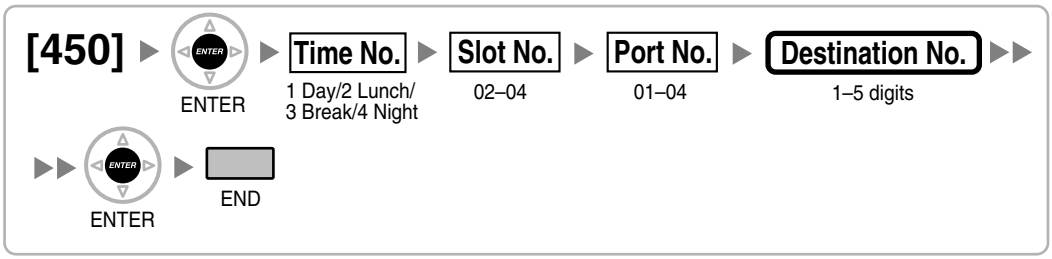
This programme needs Slot Card Reset [902] to enable after setting.

DIL 1:1 Destination [450]

[KX-TDA100/KX-TDA200]

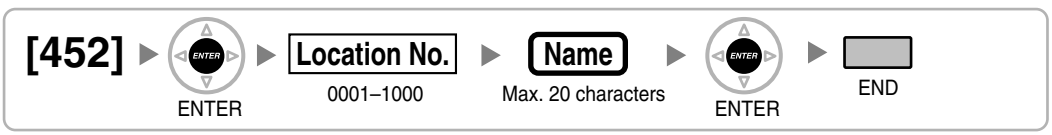


[KX-TDA30]

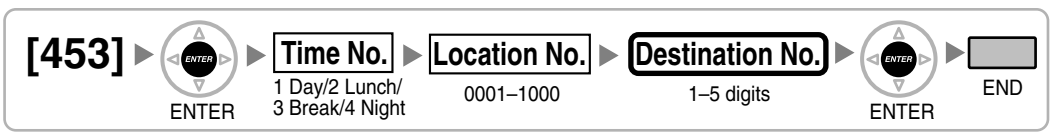


DID Number [451]

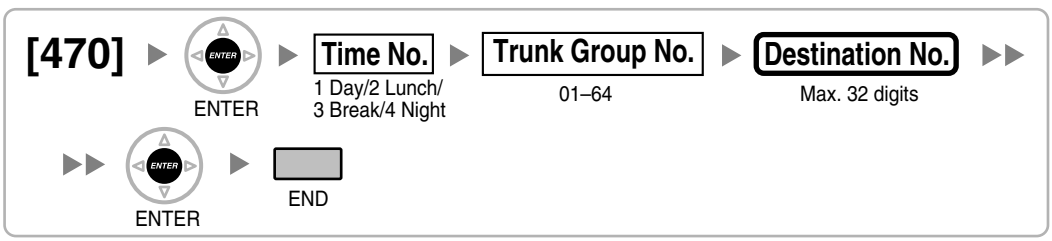
DID Name [452]



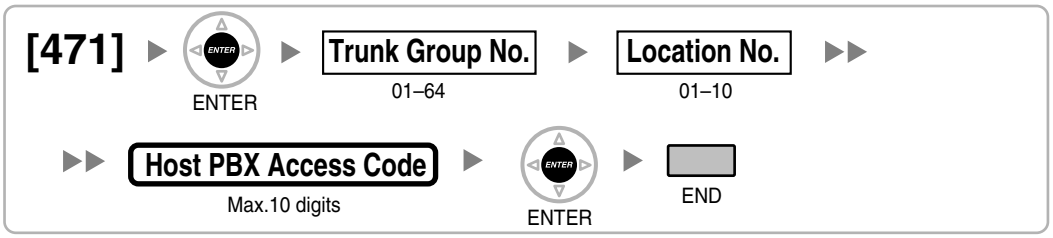
DID Destination [453]



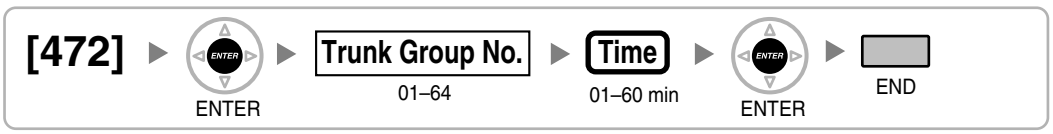
Trunk Group Intercept Destination [470]



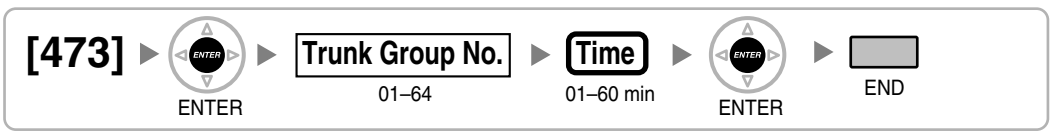
Host PBX Access Code [471]



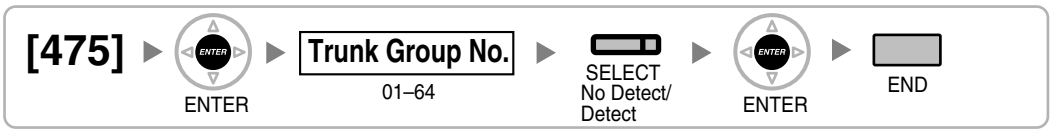
Extension-to-Trunk Call Duration [472]



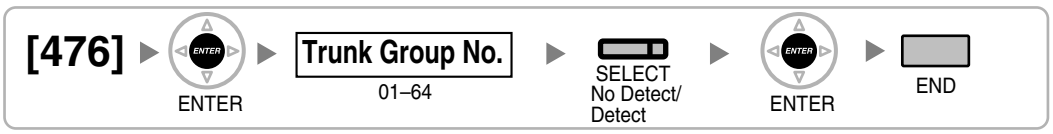
Trunk-to-Trunk Call Duration [473]



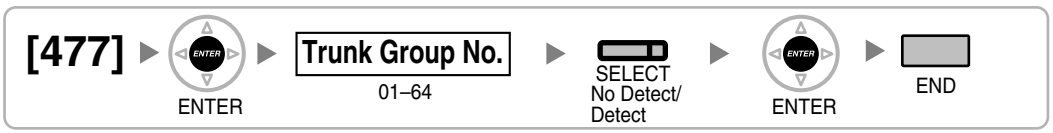
DISA Silence Detection [475]



DISA Continuous Signal Detection [476]

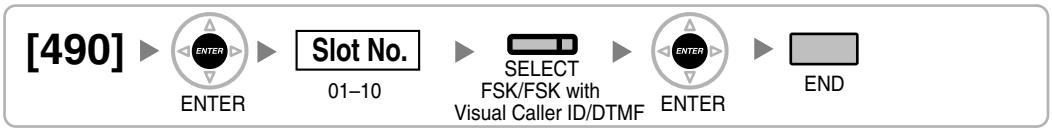


DISA Cyclic Signal Detection [477]

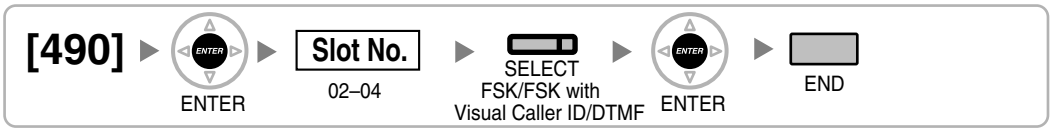


Caller ID Signal Type [490]

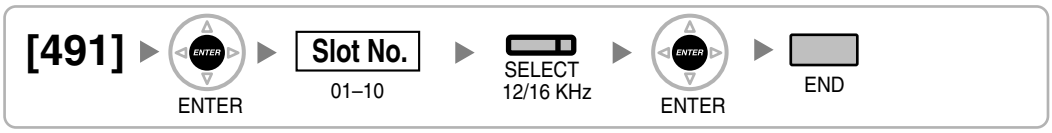
[KX-TDA100/KX-TDA200]



[KX-TDA30]

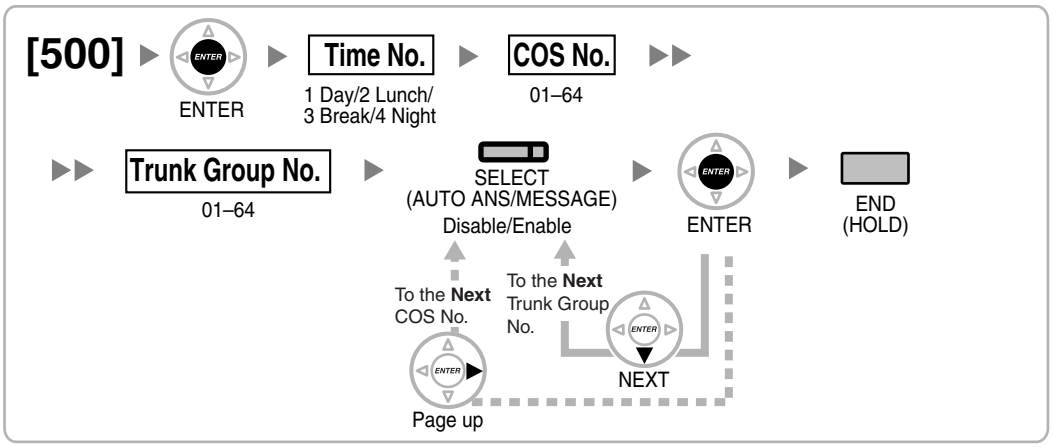


Pay Tone Signal Type [491] (KX-TDA100/KX-TDA200 only)

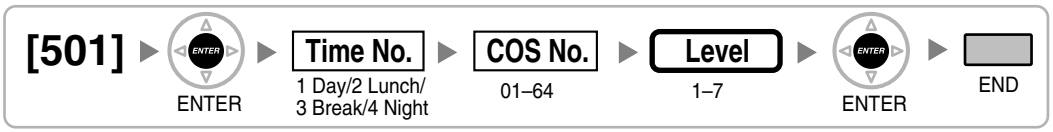


3.3.7 COS Programming

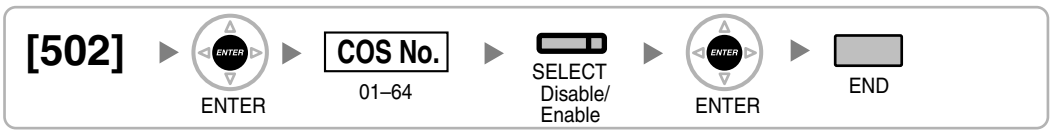
Trunk Group Number [500]



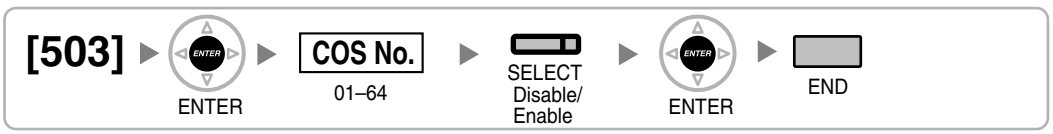
TRS/Barring Level [501]



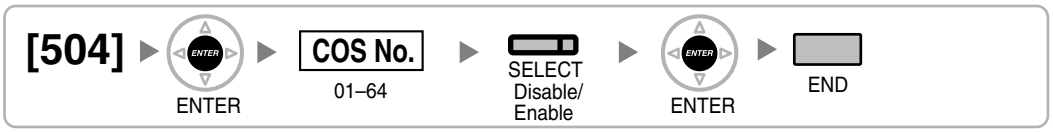
Trunk Call Duration Limitation [502]



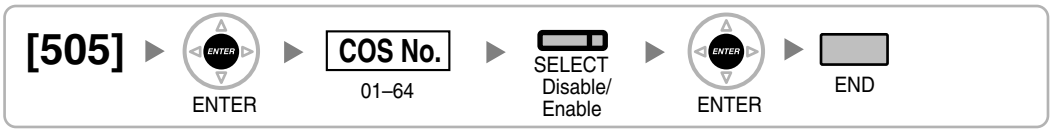
Call Transfer to Trunk [503]



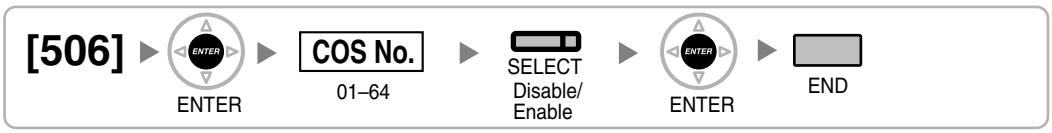
Call Forwarding to Trunk [504]



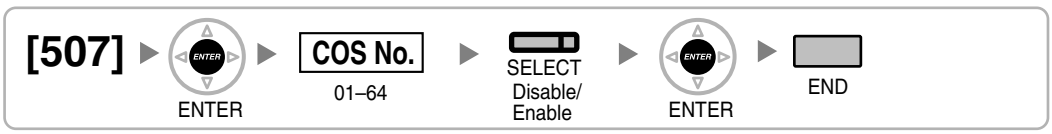
Executive Busy Override [505]



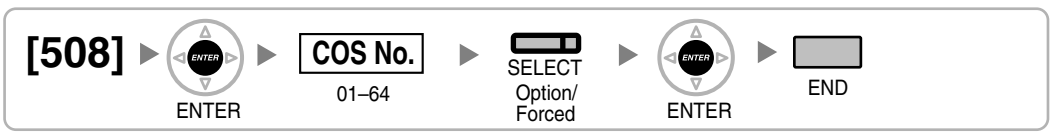
Executive Busy Override Deny [506]



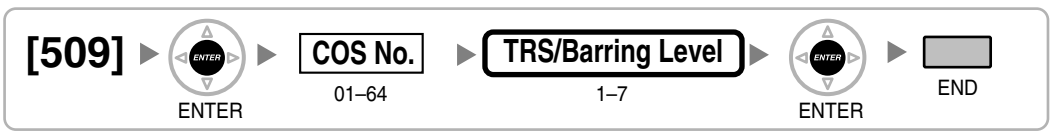
DND Override [507]



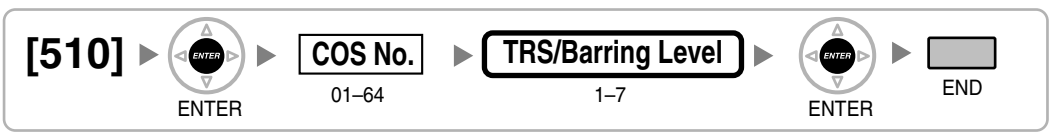
Account Code Mode [508]



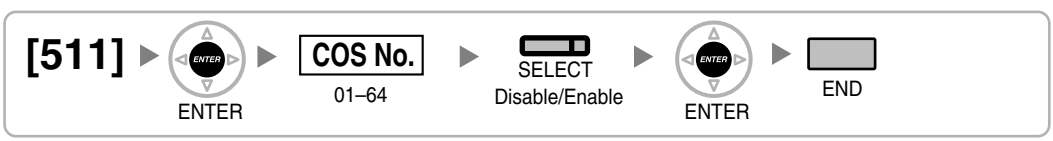
TRS/Barring Level for System Speed Dialling [509]



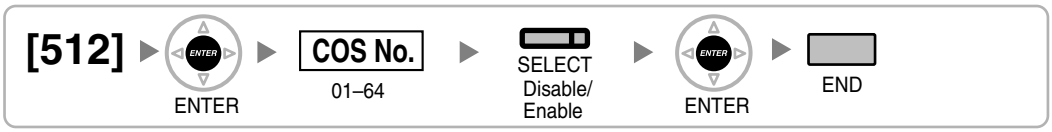
TRS/Barring Level for Extension Lock [510]



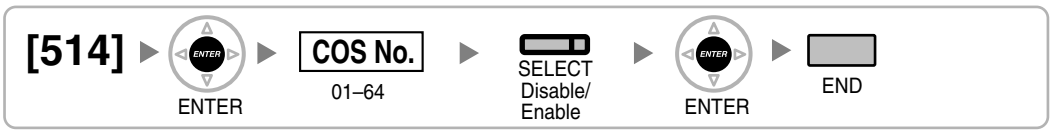
Manager Assignment [511]



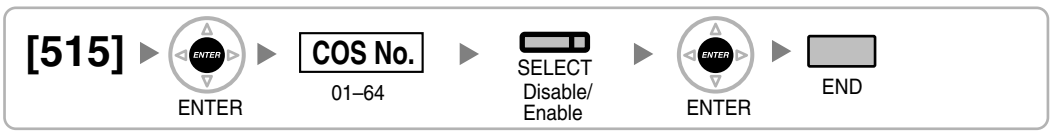
Permission for Door Open Access [512]



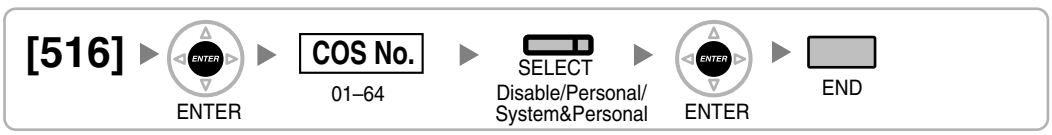
Time Service Manual Switching [514]



Wireless XDP Parallel Mode for Paired Telephone [515]



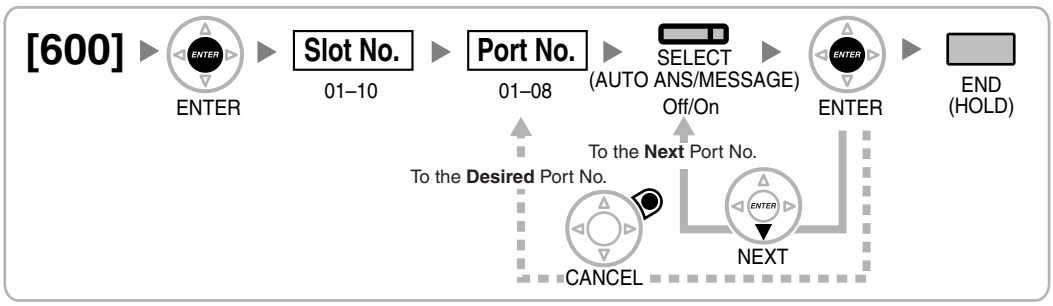
Programming Mode Limitation [516]



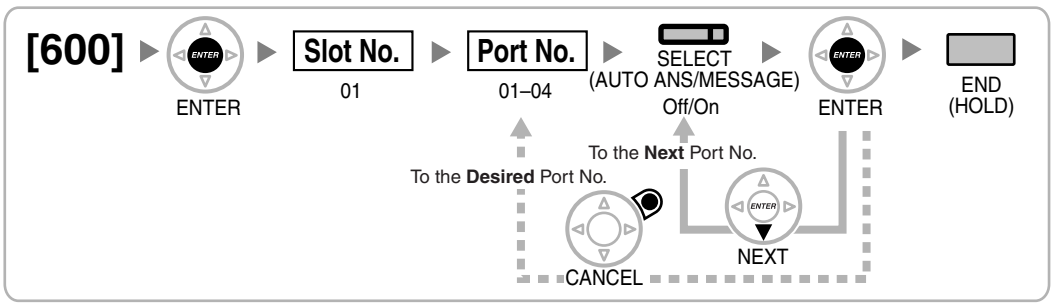
3.3.8 Extension Programming

EXtra Device Port (XDP) Mode [600]

[KX-TDA100/KX-TDA200]

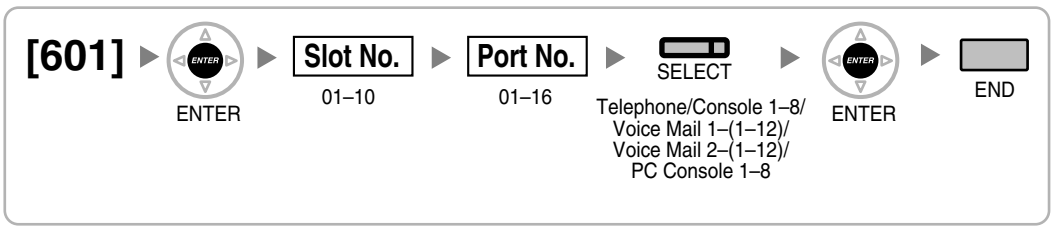


[KX-TDA30]

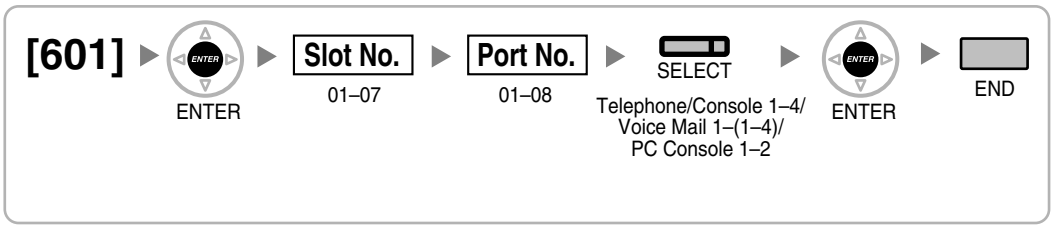


Terminal Device Assignment [601]

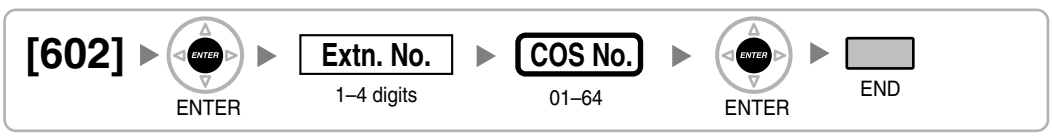
[KX-TDA100/KX-TDA200]



[KX-TDA30]

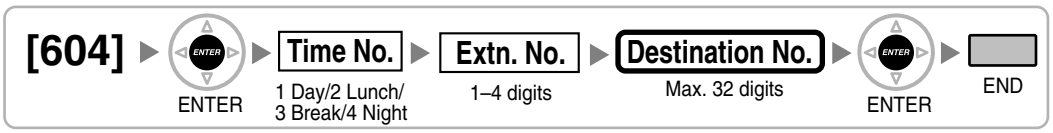


Class of Service [602]

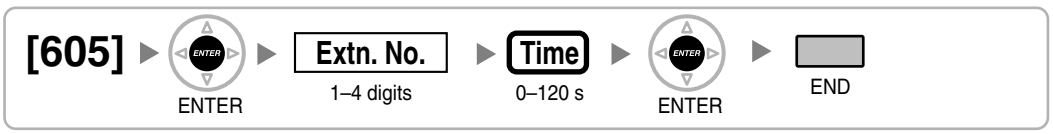


User Group [603]

Extension Intercept Destination [604]

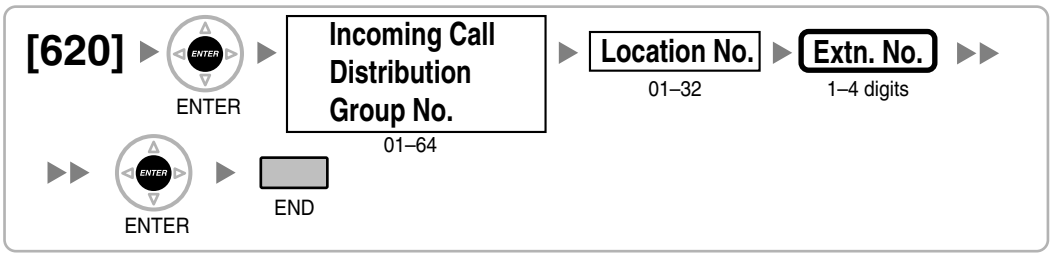


Call Forwarding—No Answer Time [605]

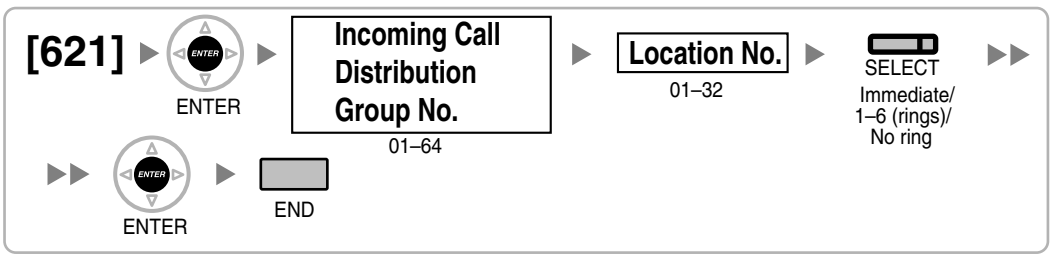


CLIP/COLP Number [606]

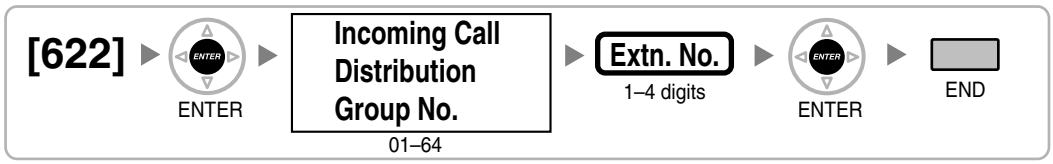
Incoming Call Distribution Group Member [620]



Incoming Call Distribution Group Delayed Ringing [621]

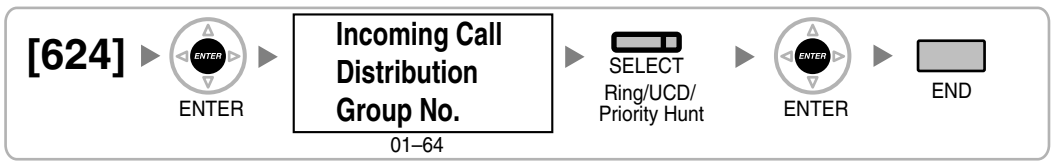


Incoming Call Distribution Group Floating Extension Number [622]

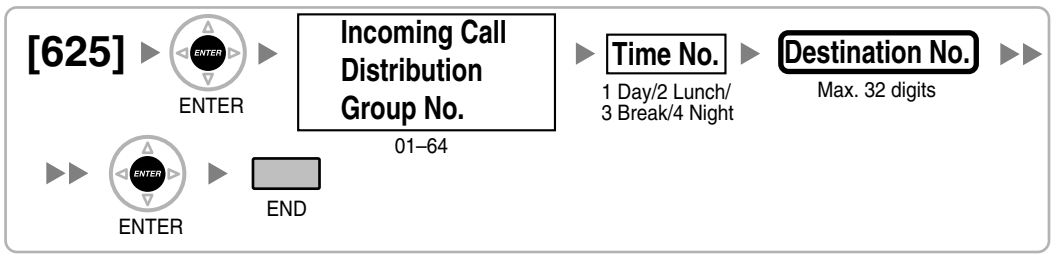


Incoming Call Distribution Group Name [623]

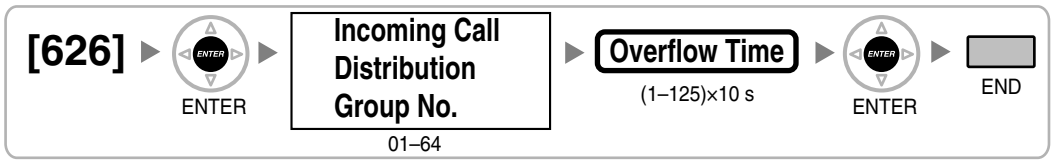
Incoming Call Distribution Group Distribution Method [624]



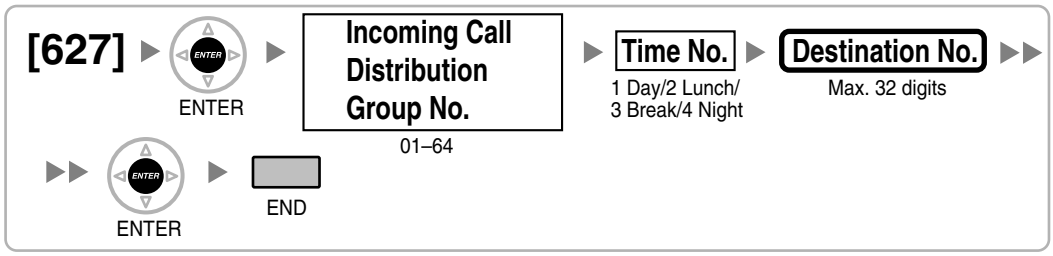
Destination for Overflow Time Expiration [625]



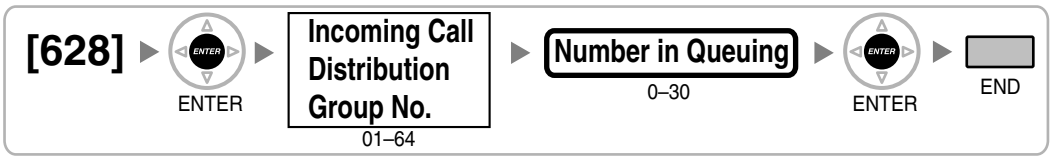
Overflow Time [626]



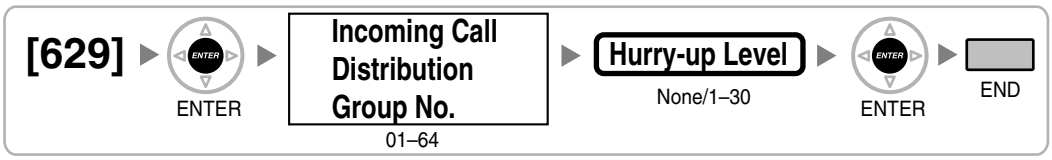
Destination When All Busy [627]



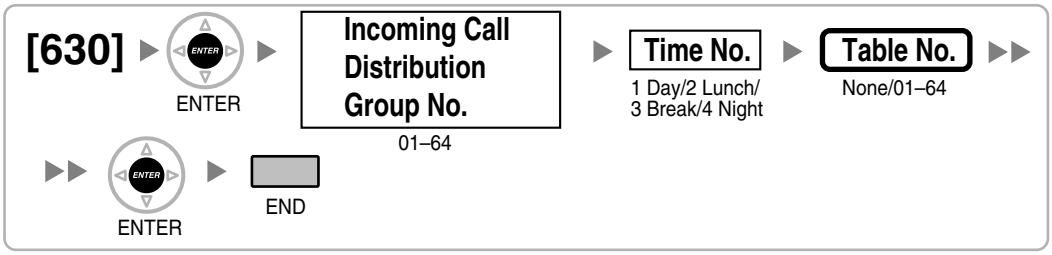
Queuing Call Capacity [628]



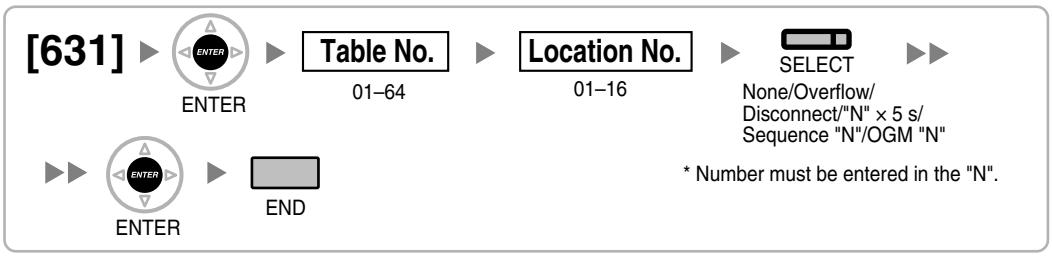
Queuing Hurry-up Level [629]



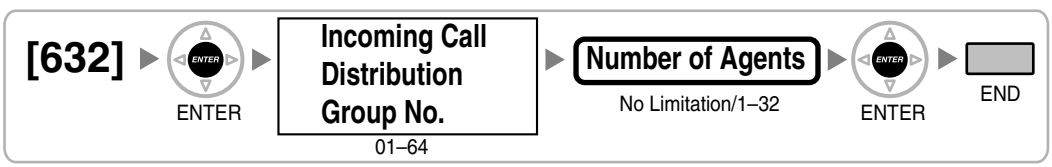
Queuing Time Table [630]



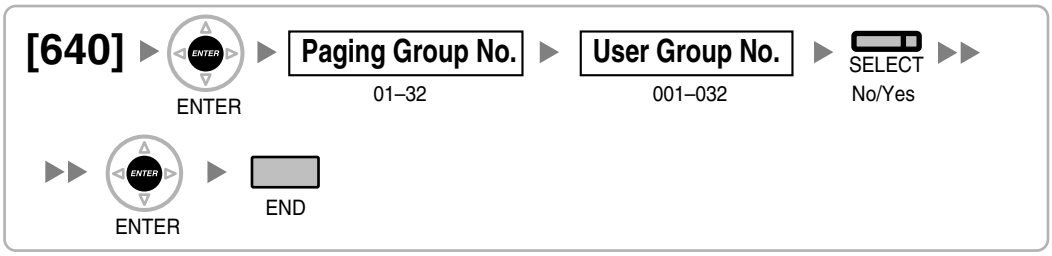
Sequences in Queuing Time Table [631]



Maximum Number of Agents [632]

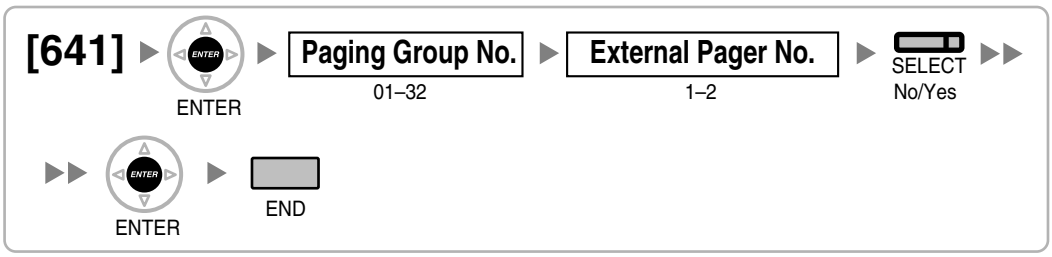


User Groups of a Paging Group [640]

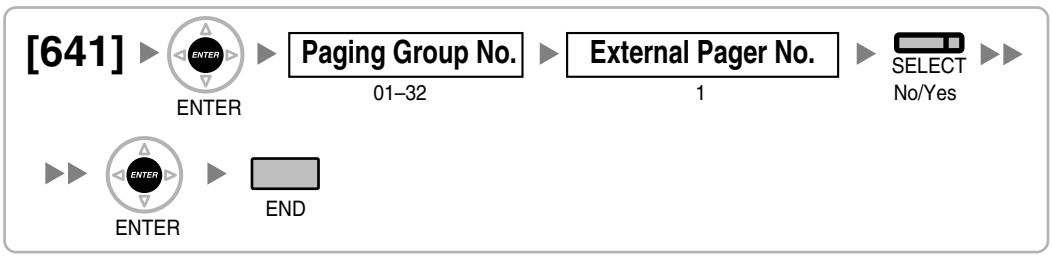


External Pagers of a Paging Group [641]

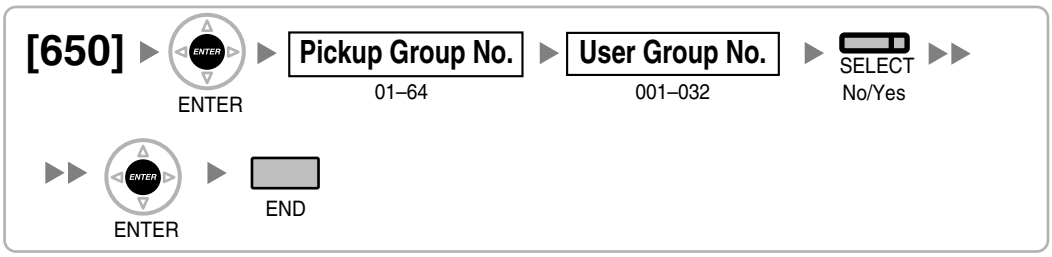
[KX-TDA100/KX-TDA200]



[KX-TDA30]



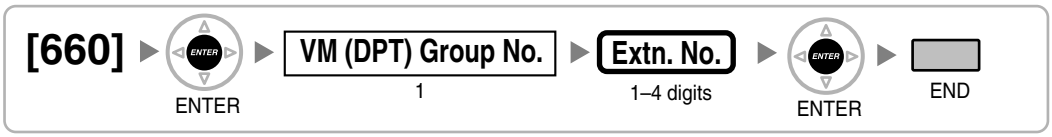
User Groups of a Pickup Group [650]



VM Group Floating Extension Number [660]

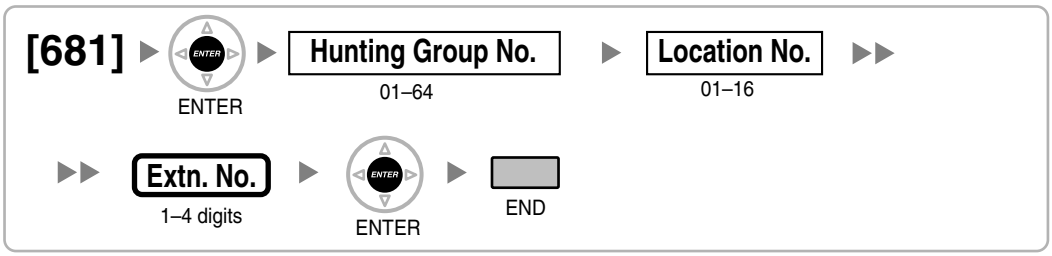
[KX-TDA100/KX-TDA200]

[KX-TDA30]



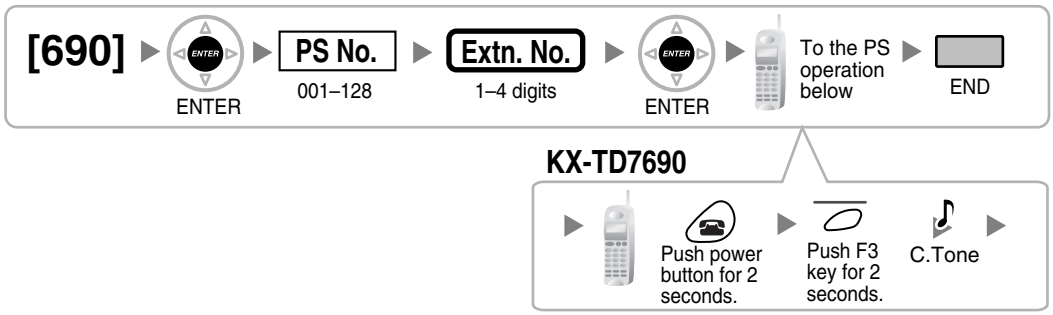
Idle Extension Hunting Type [680]

Idle Extension Hunting Group Member [681]

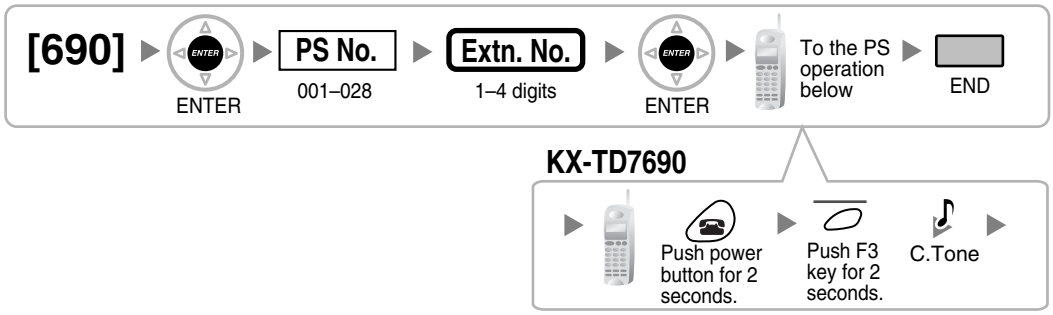


PS Registration [690]

[KX-TDA100/KX-TDA200]



[KX-TDA30]



Note

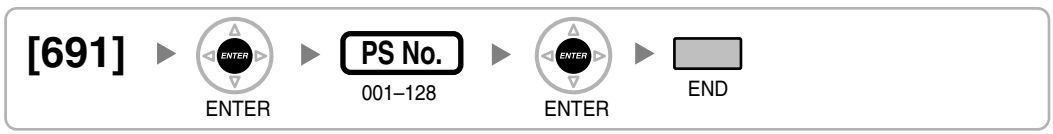
Please refer to Installation Manual for the PS registration in detail.

With the KX-TDA100/KX-TDA200:
For registering the DECT PS → 2.8.7 Connecting the Cell Station to the Hybrid IP-PBX
For registering the 2.4 GHz PS → 2.9.7 Connecting the Cell Station to the Hybrid IP-PBX

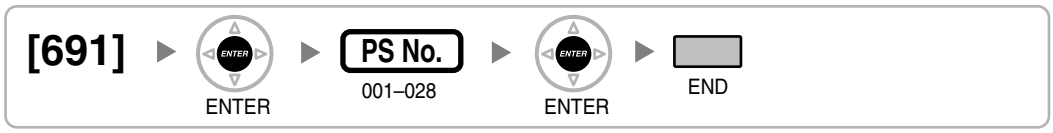
With the KX-TDA30:
For registering the DECT PS → 6.7.7 Connecting the Cell Station to the Hybrid IP-PBX
For registering the 2.4 GHz PS → 6.8.7 Connecting the Cell Station to the Hybrid IP-PBX

PS Termination [691]

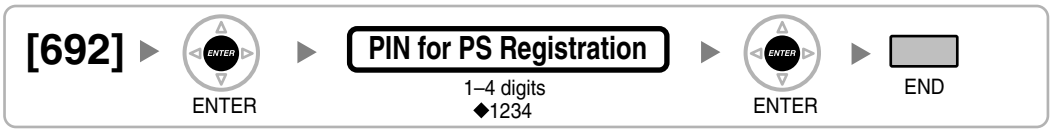
[KX-TDA100/KX-TDA200]



[KX-TDA30]



Personal Identification Number (PIN) for PS Registration [692]



CS Status Reference [699]

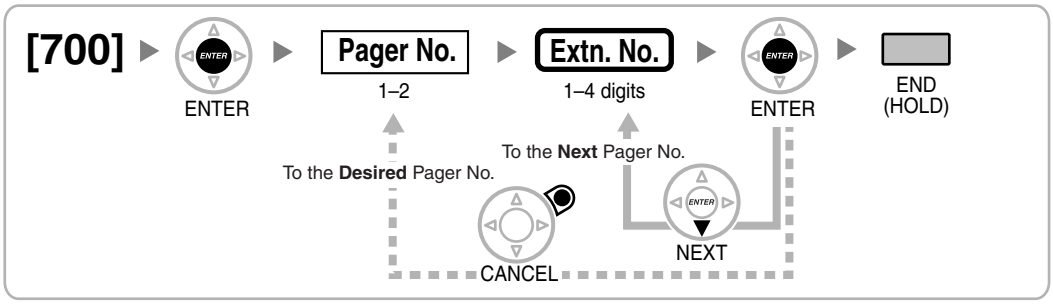
[KX-TDA100/KX-TDA200]

[KX-TDA30]

3.3.9 Resource/Interface Programming

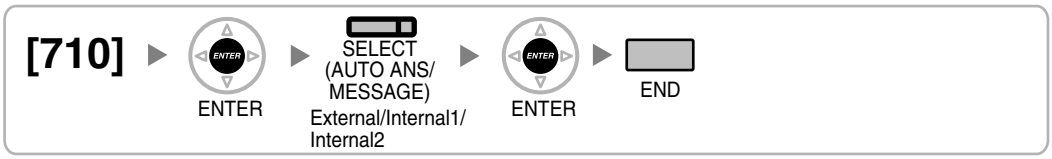
External Pager Floating Extension Number [700]

[KX-TDA100/KX-TDA200]



[KX-TDA30]

Music Source Selection for BGM (with the KX-TDA30)/BGM2 (with the KX-TDA100/KX-TDA200) [710]

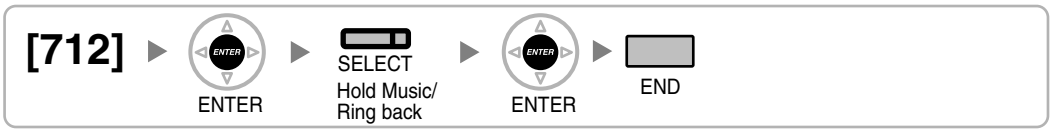


Music on Hold [711]

[KX-TDA100/KX-TDA200]

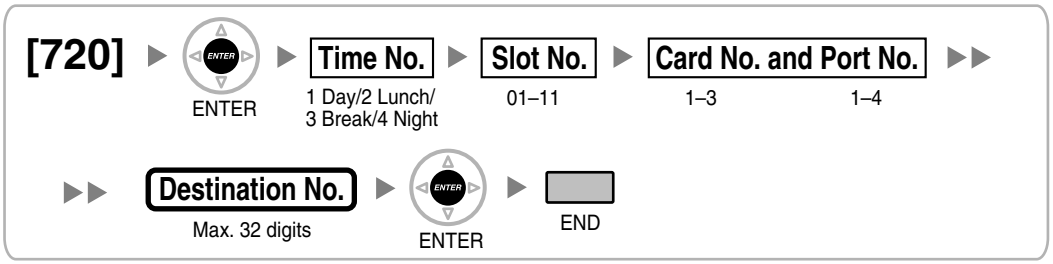
[KX-TDA30]

Music for Transfer [712]

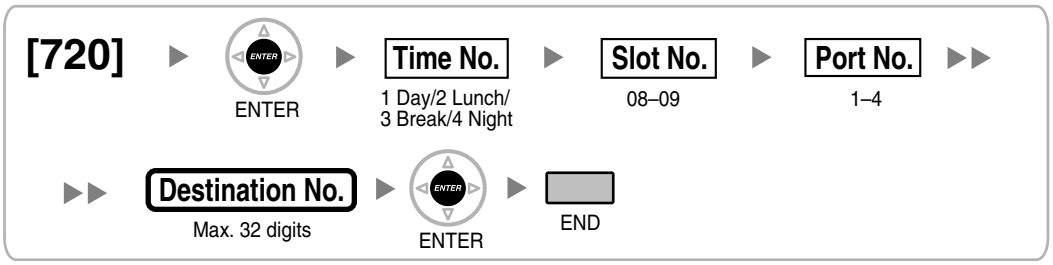


Doorphone Call Destination [720]

[KX-TDA100/KX-TDA200]

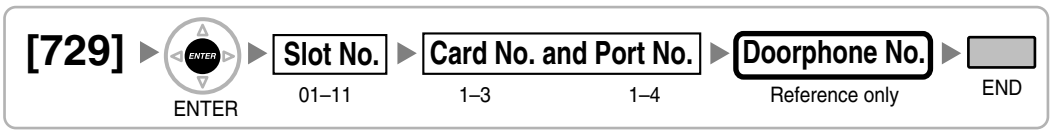


[KX-TDA30]



Doorphone Number Reference [729]

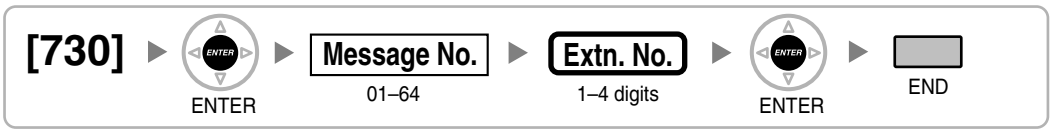
[KX-TDA100/KX-TDA200]



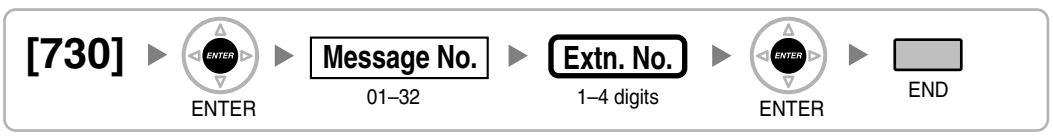
[KX-TDA30]

Outgoing Message (OGM) Floating Extension Number [730]

[KX-TDA100/KX-TDA200]



[KX-TDA30]



Outgoing Message (OGM) Name [731]

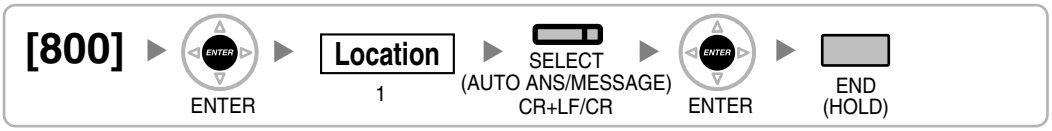
[KX-TDA100/KX-TDA200]

[KX-TDA30]

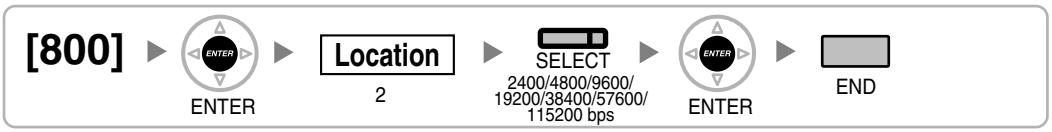
DISA Security Mode [732]

3.3.10 SMDR & Maintenance Programming

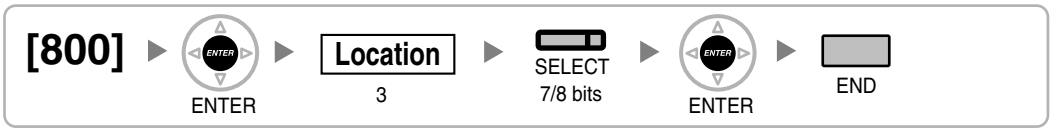
RS-232C Parameter—New Line Code [800]



RS-232C Parameter—Baud Rate [800]

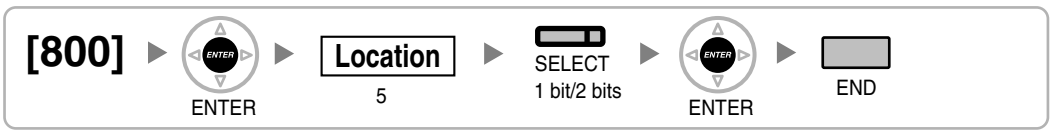


RS-232C Parameter—Word Length [800]



RS-232C Parameter—Parity Bit [800]

RS-232C Parameter—Stop Bit Length [800]



External Modem Control [801]

SMDR Page Length [802]

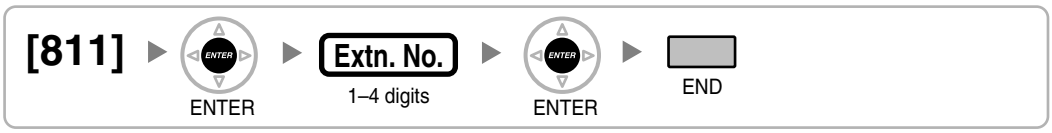
SMDR Skip Perforation [803]

SMDR Outgoing Call Printing [804]

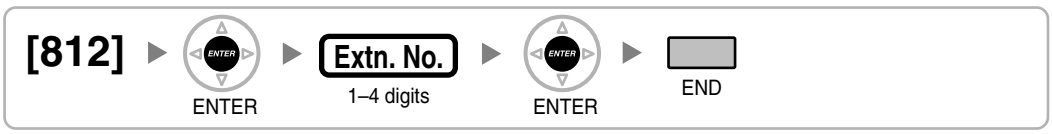
SMDR Incoming Call Printing [805]

Remote Programming [810]

Modem Floating Extension Number [811]



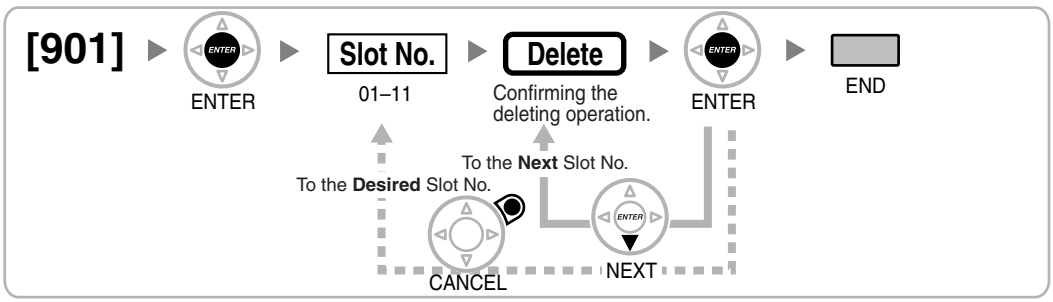
ISDN Remote Floating Extension Number [812]



3.3.11 Card Programming

Slot Card Type Reference [900]

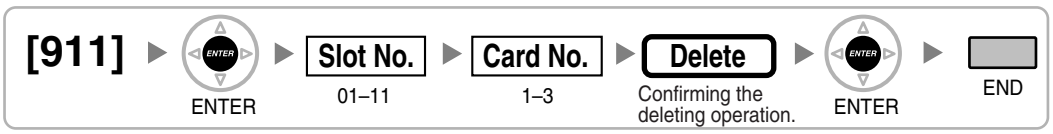
Slot Card Deletion [901]



Slot Card Reset [902]

OPB3 Option Card Type Reference [910] (KX-TDA100/KX-TDA200 only)

OPB3 Option Card Deletion [911] (KX-TDA100/KX-TDA200 only)



Section 4
Appendix

4.1 Capacity of System Resources

4.1.1 Capacity of System Resources

Category	Item	KX-TDA30	KX-TDA100	KX-TDA200
System	Tenant	8	8	8
	COS	64	64	64
	Trunk Group	64	64	64
	User Group	32	32	32
	Paging Group	32	32	32
	Call Pickup Group	64	64	64
	Incoming Call Distribution Group	64 (32 extensions/ group)	64 (32 extensions/ group)	64 (32 extensions/ group)
	PS Ring Group	32	32	32
	VM (DPT) Group	1 VPS × 8 ch	2 VPSs × 24 ch	2 VPSs × 24 ch
	VM (DTMF) Group	2 groups × 24 ch	2 groups × 32 ch	2 groups × 32 ch
	Queuing Time Table	64	64	64
	Idle Extension Hunting Group	64 (16 extensions/ group)	64 (16 extensions/ group)	64 (16 extensions/ group)
	Absent Message—System	8 × 16 characters	8 × 16 characters	8 × 16 characters
	Absent Message—Extension	1 × 16 characters/ extn.	1 × 16 characters/ extn.	1 × 16 characters/ extn.
	Message Waiting	256	256	256
	Number of Characters of Name	20	20	20
	Extension Number Digits	1 – 4	1 – 4	1 – 4
	Extension Personal Identification Number (PIN)	10 digits/extn.	10 digits/extn.	10 digits/extn.
	Call Park Zone	100	100	100
	Conference	3 × 10 – 8 × 4	3 × 10 – 8 × 4	3 × 10 – 8 × 4
	Verified Code	4 digits, 1 000 entries	4 digits, 1 000 entries	4 digits, 1 000 entries
	Verified Code Personal Identification Number (PIN)	10 digits, 1 000 entries	10 digits, 1 000 entries	10 digits, 1 000 entries

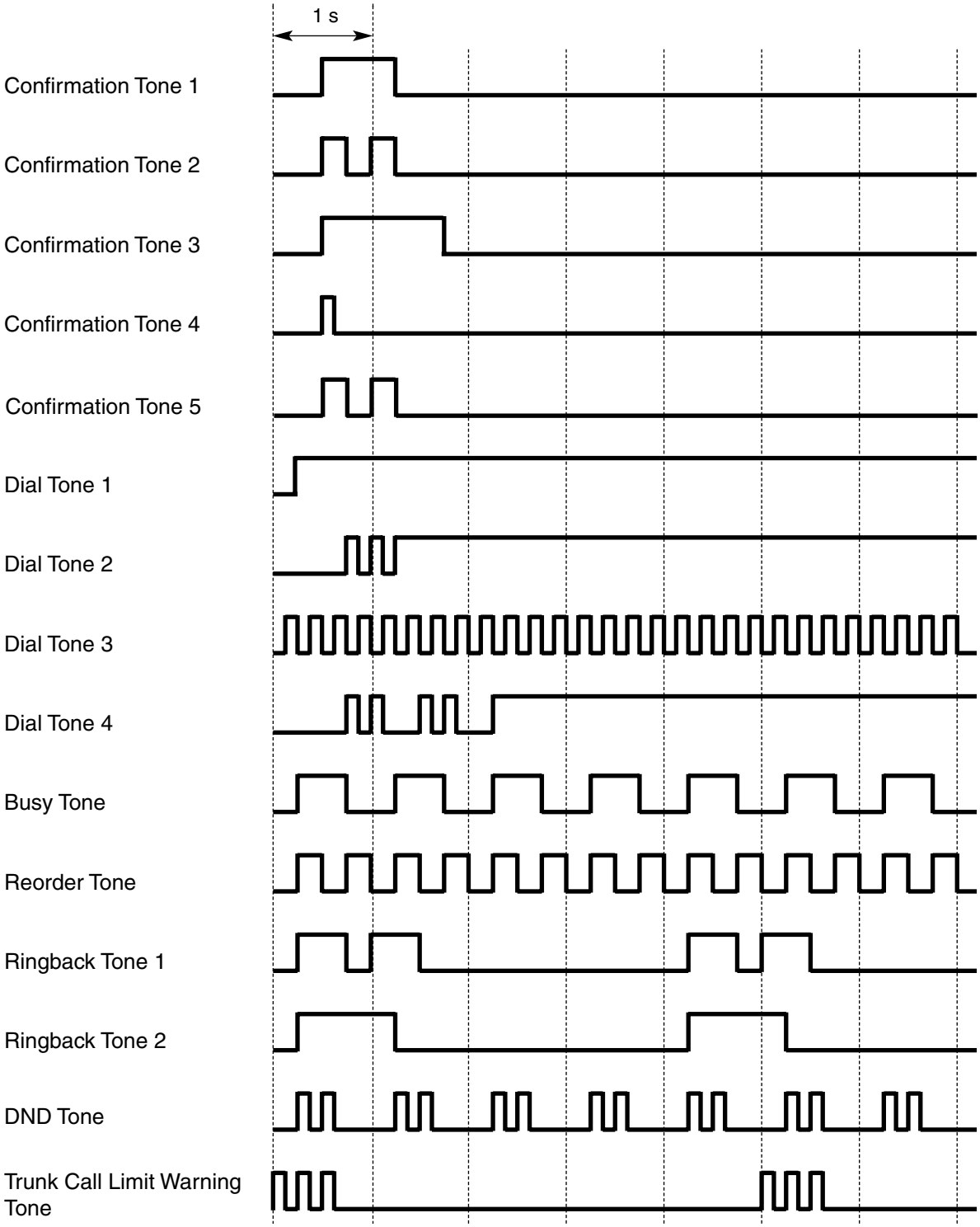
Category	Item	KX-TDA30	KX-TDA100	KX-TDA200
System	Special Carrier Code	16 digits, 20 entries	16 digits, 20 entries	16 digits, 20 entries
	Host PBX Access Code	10 digits, 10 entries/trunk group	10 digits, 10 entries/trunk group	10 digits, 10 entries/trunk group
	DDI/DID Table	32 digits, 1 000 entries	32 digits, 1 000 entries	32 digits, 1 000 entries
	SMDR	200 calls	200 calls	200 calls
Dialling	Emergency Call	32 digits, 10 entries	32 digits, 10 entries	32 digits, 10 entries
	Quick Dialling	4 digits, 80 entries	4 digits, 80 entries	4 digits, 80 entries
	System Speed Dialling	32 digits, 1 000 entries	32 digits, 1 000 entries	32 digits, 1 000 entries
	Personal Speed Dialling	32 digits, 10 entries/extn.	32 digits, 10 entries/extn.	32 digits, 10 entries/extn.
	One-touch Dialling	32 digits	32 digits	32 digits
	Hot Line	32 digits	32 digits	32 digits
	Key Pad Protocol Dial (ISDN Service Access)	32 digits	32 digits	32 digits
	Redial	32 digits	32 digits	32 digits
ARS	Routing Plan Table	16 entries	16 entries	16 entries
	Leading Number Table	16 digits, 1 000 entries	16 digits, 1 000 entries	16 digits, 1 000 entries
	Leading Number Exception Table	16 digits, 200 entries	16 digits, 200 entries	16 digits, 200 entries
	ARS Carrier	10	10	10
	Itemised Billing Code	10 digits	10 digits	10 digits
	Authorisation Code	10 digits	10 digits	10 digits
Networking	TIE Line Routing and Modification Table	32 entries	32 entries	32 entries
	Leading Number	3 digits	3 digits	3 digits
	PBX Code	7 digits	7 digits	7 digits
TRS/Barring	TRS/Barring Level	7	7	7
	TRS/Barring Denied Code	16 digits, 100 entries/level	16 digits, 100 entries/level	16 digits, 100 entries/level
	TRS/Barring Exception Code	16 digits, 100 entries/level	16 digits, 100 entries/level	16 digits, 100 entries/level

Category	Item	KX-TDA30	KX-TDA100	KX-TDA200
Call Log	Outgoing Call Log—PT	100 logs/extn. 5 × 28 logs/PBX	100 logs/extn. 5 × 128 logs/PBX	100 logs/extn. 5 × 128 logs/PBX
	Outgoing Call Log—PS	100 logs/extn. 5 × 28 logs/PBX	100 logs/extn. 5 × 128 logs/PBX	100 logs/extn. 5 × 128 logs/PBX
	Incoming Call Log—PT	100 logs/extn. 10 × 28 logs/PBX	100 logs/extn. 10 × 128 logs/PBX	100 logs/extn. 10 × 128 logs/PBX
	Incoming Call Log—PS	100 logs/extn. 10 × 28 logs/PBX	100 logs/extn. 10 × 128 logs/PBX	100 logs/extn. 10 × 128 logs/PBX
	Incoming Call Log—Incoming Call Distribution Group	100 logs/group 10 × 64 logs/PBX	100 logs/group 10 × 64 logs/PBX	100 logs/group 10 × 64 logs/PBX
Password	System Password for Installer	4 – 10 characters	4 – 10 characters	4 – 10 characters
	System Password for Administrator—for PC Programming	4 – 10 characters	4 – 10 characters	4 – 10 characters
	System Password for User—for PC Programming	4 – 10 characters	4 – 10 characters	4 – 10 characters
	System Password for Administrator—for PT Programming	4 – 10 digits	4 – 10 digits	4 – 10 digits
	System Password for User—for PT Programming	4 – 10 digits	4 – 10 digits	4 – 10 digits
	Manager Password	4 – 10 digits	4 – 10 digits	4 – 10 digits
Programmer Code	Installer Level Programmer Code	4 – 16 characters	4 – 16 characters	4 – 16 characters
	Administrator Level Programmer Code	4 – 16 characters	4 – 16 characters	4 – 16 characters
	User Level Programmer Code	0 – 16 characters	0 – 16 characters	0 – 16 characters

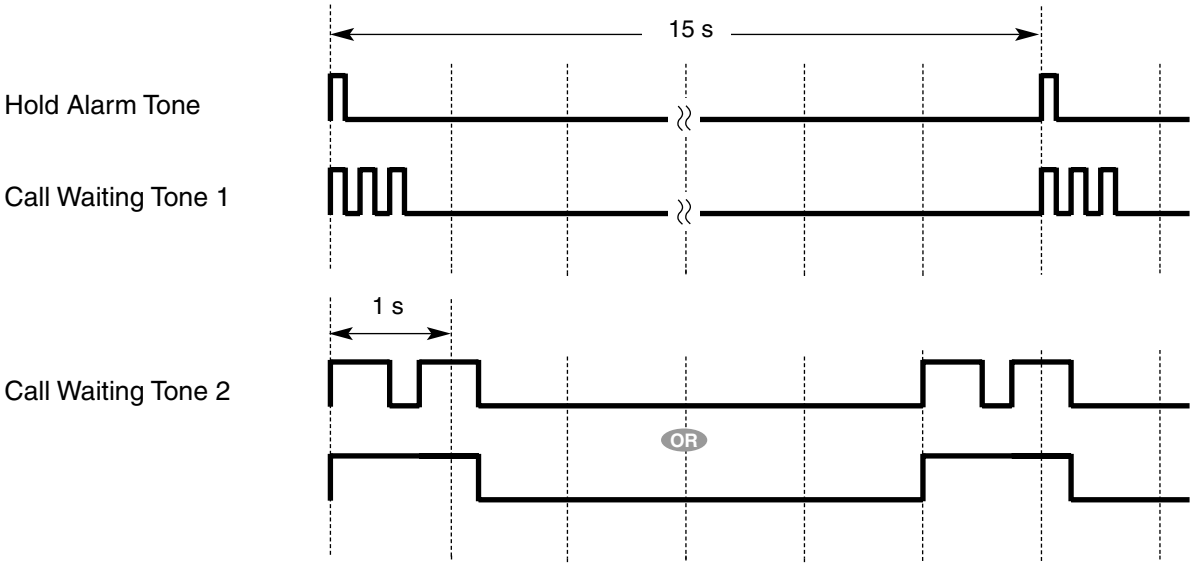
4.2 Tones/Ring Tones

4.2.1 Tones/Ring Tones

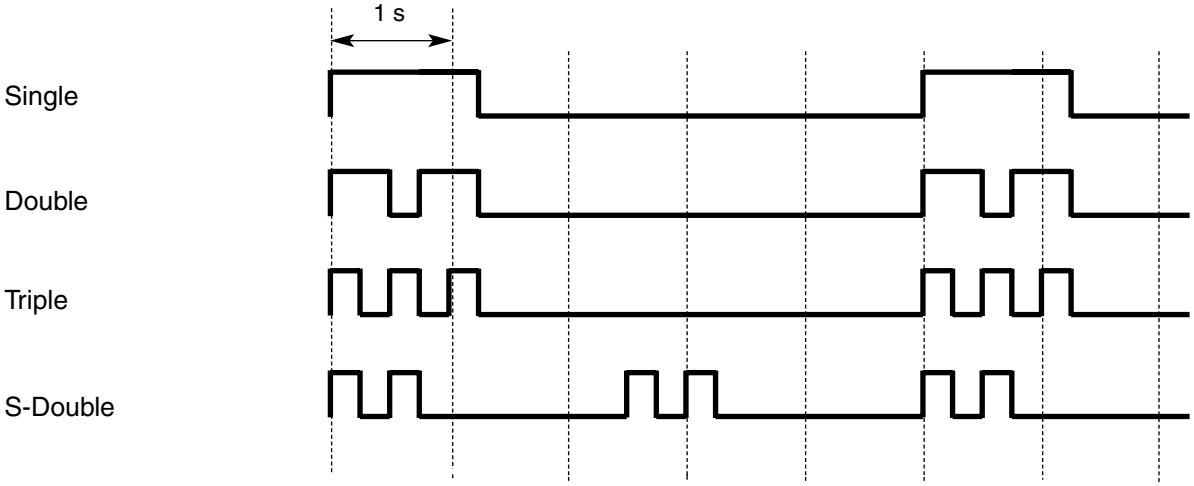
Tone Patterns (Default)



Tone Patterns (Default)



Ring Tone Patterns (Default)



4.3 List of Abbreviations

4.3.1 List of Abbreviations

A

AA → Automated Attendant
ANI → Automatic Number Identification
AOC → Advice of Charge
APT → Analogue Proprietary Telephone
ARS → Automatic Route Selection

B

BGM → Background Music
BRI → Basic Rate Interface

C

CCBS → Completion of Calls to Busy Subscriber
CF → Call Forwarding—by ISDN
CLI → Calling Line Identification
CLIP → Calling Line Identification Presentation
CLIR → Calling Line Identification Restriction
CNIP → Calling Name Identification Presentation
CNIR → Calling Name Identification Restriction
COLP → Connected Line Identification Presentation
COLR → Connected Line Identification Restriction
CONP → Connected Name Identification Presentation
CONR → Connected Name Identification Restriction
COS → Class of Service
CPC → Calling Party Control
CS → Cell Station
CT → Call Transfer—by ISDN
CTI → Computer Telephony Integration

D

DDI → Direct Dialling In
DID → Direct Inward Dialling
DIL → Direct In Line
DISA → Direct Inward System Access
DND → Do Not Disturb
DPT → Digital Proprietary Telephone
DSS → Direct Station Selection
DTMF → Dual Tone Multi-Frequency

E

EFA → External Feature Access

F

FWD → Call Forwarding

G

G-CO → Group-CO

I

ICD → Incoming Call Distribution
IRNA → Intercept Routing—No Answer
ISDN → Integrated Services Digital Network

L

L-CO → Loop-CO
LCS → Live Call Screening
LED → Light Emitting Diode

M

MCID → Malicious Call Identification
MSN → Multiple Subscriber Number

O

OGM → Outgoing Message
OHCA → Off-hook Call Announcement
OPX → Off Premise Extension

P

P-MP → Point-to-multipoint
P-P → Point-to-Point
PIN → Personal Identification Number
PRI → Primary Rate Interface
PS → Portable Station
PT → Proprietary Telephone

S

S-CO → Single-CO
SLT → Single Line Telephone
SMDR → Station Message Detail Recording

T

TAFAS → Trunk Answer from Any Station
TEI → Terminal Endpoint Identifier
TRG → Trunk Group
TRS/Barring → Toll Restriction/Call Barring

U

UCD → Uniform Call Distribution

V

VM → Voice Mail
VoIP → Voice over Internet Protocol
VPN → Virtual Private Network
VPS → Voice Processing System

X

XDP → eXtra Device Port

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